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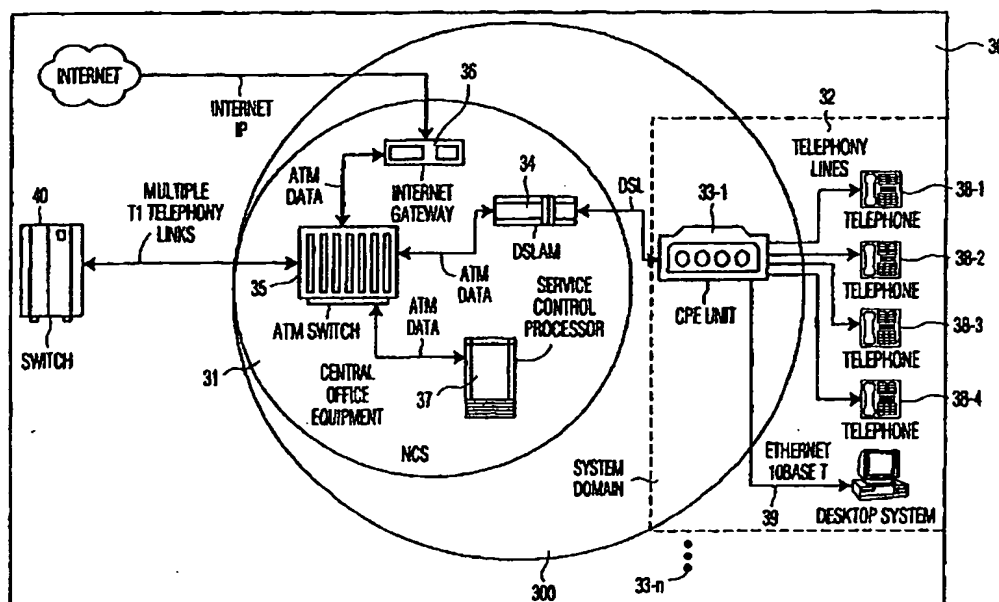
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(54) Title: SYSTEM AND METHOD FOR PROVIDING VOICE AND/OR DATA SERVICES



(57) Abstract: A system and method for providing a voice and/or data service is thus presented, comprising the following. A connection request is detected. The connection request is formatted into a data cell. The data cell is sent to a service control point. The service control point determines in real time whether a connection to public switch network is available. If the connection to a public switch network is available, the requested connection is set up as a virtual circuit connection to the public switch network.

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SYSTEM AND METHOD FOR PROVIDING VOICE AND/OR DATA SERVICESINTRODUCTION

5       The present invention generally relates to a communication system and method, and more particularly to a system and method of providing voice and/or data services using, for example, a DSL communication system.

BACKGROUND OF THE INVENTION

10       With the increasing popularity of the Internet, more and more business and residential customers are demanding high-speed broadband access. Broadband access may be provided via different types of physical medium, such as cable, satellite or traditional copper, telephone wires.

15       Since telephone wires typically represent high percentage of penetration in existing homes and businesses, various innovations have been proposed to increase the bandwidth of the existing telephone local plant. One such innovation is, for example, a modem which may operate with a speed of up to 56 kbps. Another  
20       innovation is Digital Subscriber Line (DSL) service. DSL service uses existing copper local loop for providing services to an end user. Another attraction of DSL is that it is capable of providing both high-speed data service and telephony voice service using the same telephone wire pair.

25       A generic network topology for DSL is illustrated in Fig. 1. DSL service is delivered over regular telephone wires 2-1 or 2-2 using DSL Access Multiplexers (DSLAMs) 3 in a central office. For customers who receive only data service over DSL, the service is terminated at the customer premises with a DSL modem or router, shown as CPE  
30       equipment 4. In a traditional system offering DSL data only services, the POTS service coexists with the DSL service since POTS signaling is in baseband and DSL exists in a higher frequency band on the same wire coming to/from the customer premises. At both the customer premises site and the central office, POTS splitters 15 and 17 function  
35       to split the POTS signal from the data signal. The POTS signal is routed to the traditional voice network 14 at the central office and DSL data is routed to DSLAM 3, as shown in Fig. 1.

40       The next generation of devices have attempted to enhance the value of the DSL link by adding value added services such as digital voice links on this network. In such a model, digitized (and possibly compressed for bandwidth efficiency) voice is carried along with data

## 2

over the DSL link. For combined voice and data service, DSL service is terminated by a device that provides integrated voice and data access, shown as CPE 5 in Fig. 1. Such devices typically offer an Ethernet port 6 for data and multiple analog POTS ports, for example, 7-1 and 7-2 for voice.

A DSLAM 3 serves as a packet multiplexer, typically delivering traffic from multiple customers over a high-speed uplink 8 to a metropolitan or regional packet network 9. The data network typically consists of an ATM switch. The packet protocol uses by the DSLAMs are packet protocols such as ATM or frame relay to support voice and data. Since the main data service used by DSL customers is accessing the Internet 11, the packet network is connected to the Internet, typically through a device known as a Subscriber Management System 12. Connection to an intra or enterprise data networks may also be used, to support home-based workers, for example.

A voice gateway 13 is used to deliver voice services to DSL customers. The voice gateway 13 connects the packet network to the public switched telephone network (PSTN) 14. Digital voice streams are converted into packet format for transport over the packet network between the voice gateway and the integrated access device on the customer premises. The voice gateway connects to the PSTN via a telephone switch, for example, a Class 4 or 5 switch.

Since the voice gateway 13 represents a digital access network from the point of view of the telephone switch, the connection between the gateway and the telephone switch typically makes use of a standard interface for digital loop carrier system such as GR-303, TR-008 or V5. This class of signaling is known as in-band signaling. There is a second class of signaling employed to communicate with a more intelligent digital loop carrier or edge device. This signaling stack is known as SS7 and represents an out-of-band signaling paradigm. The connection of the edge device to the telephone through the SS7 protocol stack is a link that is logically separate from the synchronous timeslots used in the class of in-band signaling protocols. What has been described so far is the known generic architecture for implementing a DSL service.

In addition to the traditional data and POTS services, various equipment manufacturers are introducing equipment for integrated digital voice and data services over DSL. As an example, Coppercom Communications Inc., of California provides equipment to DSL service providers for offering DSL services. In particular, the company has proposed a CopperComplete™ DSL architecture, shown for example, in Fig. 2.

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The system architecture provided by CopperComplete™ DSL uses a voice gateway 21 behind the ATM switch 22. The voice gateway 21 is an additional piece of equipment that converts the packetized voice traffic to voice signals acceptable to the PSTN (Public Switched Telephone Network) via a Class 5 switch 23. The voice gateway 21 converts the incoming ATM Adaptation Layer 2 (AAL2) cells to time division multiplexed voice signals and sends it to the Class 5 switch 23 using multiple T1 trunks 24. This interface is, for example, GR-303 interface, the same as used by digital loop carriers (DLC), as described before in connection with Fig. 1.

It is believed the voice path used in the Coppercom architectures is a permanent virtual circuit (PVC) that is configured during the provisioning of the CPE device, not in real time. This PVC carries all voice traffic as well as signaling traffic. The packet architecture used is ATM Adaptation Layer 2 (AAL2) for ATM encapsulation.

AAL2 has the ability to allow multiple connections multiplexed on one virtual circuit (VC). The multiplexing of multiple streams of data is done at the ATM Adaptation Layer. ATM adaptation only takes place at the endpoints of an ATM network. Cells in an ATM network are routed or switched based upon their virtual path/virtual channel (VP/VC) identifier. In the case of a permanent virtual circuit (PVC), as in the case of the Coppercom architecture, the cells are switched to the same permanent destination previously established at the time of the CPE provisioning.

The Coppercom architecture does not use the ATM network to setup and teardown the voice connections, but instead uses the voice gateway. It is, therefore, not possible to take advantage of the ATM network for switching of individual voice calls. This is because, as explained previously, in the Coppercom architecture, multiple voice calls are multiplexed along with signaling data onto a single ATM virtual circuit. The contents of the ATM cell stream are transparent to the ATM network. The ATM network only examines the header to ensure they are sent to the correct destination. The call assignment or switching in this architecture is independent of the ATM network. The call assignment cannot be determined until the signaling and voice data is de-multiplexed at the voice gateway.

## SUMMARY OF INVENTION

Present inventors recognize that there are several drawbacks to prior DSL architectures. By using ATM AAL2 to carry voice, these architectures add significant cost and complexity to the end user equipment in terms of compression (when applicable), silence suppression, variable packet fill delay settings. In addition, there will be the need to multiplexing multiple streams of data onto one

ATM virtual circuit at the CPE. Further, these architecture do not allow you to take direct advantage of the ATM network (e.g., the ATM switch) in terms of the dynamic capabilities of connection setup and teardown, resulting in inefficient use of system resources.

Therefore, present inventors have arrived at a better solution of using ATM AAL1 for carrying voice traffic, instead. AAL1 provides a simple method of carrying voice through an ATM network as well as the ability to dynamically setup and teardown connections at the ATM layer within the ATM network itself (see, e.g., *ITU-TI.363.1: B-ISDN ATM Adaptation Layer Specification: Type 1 AAL*).

Hence, one aspect of the present invention is a communication architecture whose voice path is based on a combination of a PVC and a switched virtual circuit (SVC). The PVC is setup from the CPE to the trunk port on the DSLAM. The SVC is the dynamic connection that can exist in the ATM switch for the purposes of setting up and tearing down voice connections. Each voice path is carried in an independent ATM virtual circuit, rather than multiplexing multiples of them together. The voice traffic is carried using AAL1. The signaling information is also carried independently of the voice and is also routed towards a service control processor rather than directly to the switch.

In this architecture, the value of the virtual path/virtual channel directly identifies the information contained within the cell and the user it is intended for. By using an independent ATM virtual circuit per voice channel, it is possible to setup and teardown the connection in the ATM switch (using a switched virtual circuit).

A service control processor externally controls the switch through inband signaling. This architecture exploits the power of the ATM switch by using its capabilities to dynamically setup and teardown connections. In this architecture, there is no decision making at the ATM Adaptation layer in the ATM network, the data is sent to the correct destination based upon the VP/VC. This connection is established at the ATM Layer.

A system and method for providing a voice and/or data service is thus presented, comprising the following. A connection request is detected. The connection request is formatted into a data cell. The data cell is sent to a service control point. The service control point determines in real time whether a connection to public switch network is available. If the connection to a public switch network

is available, the requested connection is set up as a virtual circuit connection to the public switch network.

#### BRIEF DESCRIPTION OF THE DRAWINGS

- 5 Fig. 1 shows a block diagram of a known DSL architecture for providing services.
- Fig. 2 shows a block diagram of another DSL architecture for providing both voice and data services.
- 10 Fig. 3 is a system diagram of a DSL system according to the principles of the present invention.
- Fig. 4 shows an exemplary detailed embodiment of a CPE unit in circuit block diagram form.
- Fig. 5 shows a block diagram for performing a downlink operation within FPGA 45.
- 15 Fig. 6 shows an uplink ATM Cell Encapsulation Interworking Function.
- Fig. 7 shows the operational scenario diagram for placing or initiating a call.
- Fig. 8 shows the scenario diagram for the call in process operational sequence.
- 20 Fig. 9 shows the scenario diagram for the CPE call disconnect operational initiated by an CPE.
- Fig. 10 shows the operational scenario diagram for incoming call.
- Fig. 11 shows the operational scenario diagram a call disconnect initiated by the telco switch or NCS.
- 25 Fig. 12 shows a block diagram of and GR-303 interface.
- Fig. 13 shows an exemplary embodiment of a SCP in distributed environment.
- Fig. 14 is a block diagram that shows how voice channels are allocated between a CPE and the Telco, through a SCP.
- 30 Fig. 15 shows the logical flow of the data paths between CPE and NCS.
- Fig. 16 is another block diagram of the data paths between the CPEs and the NCS.



Fig. 17 illustrates how voice data paths are set as variable bit rate (VBR) while the command and PC paths are set as unspecified bit rate (UBR).

Fig. 18 is an example of the data paths for voice traffic through the NCS.

Fig. 19 is an example block diagram of where throughput can be reduced for a system consisting of 6 CPEs.

### DETAILED DESCRIPTION

An exemplary embodiment according to the principles of the present invention is shown in Fig. 3.

The System Block Diagram 30 is composed of several functional blocks. The System Domain 300 is composed of Central Office (CO) Equipment domain 31 (also refers to as Network Control System (NCS) hereinafter) and Customer Premise Equipment (CPE) domain 32. The component blocks within the System Domain and their respective interfaces are:

Customer Premise Equipment (CPE) Unit 33-n

Digital Subscriber Line Access Multiplexer (DSLAM) 34

ATM Switch 35

Internet Gateway 36

Service Control Processor (SCP) 37

#### CPE Unit 33-n

The CPE unit 33-n consists of an internal DSL Modem unit that interfaces with, for example, up to 4 separate analog telephones 38-1 to 38-4 (via POTS) and one 10Base-T Ethernet connection 39. From the customer's analog end, the CPE device, for example, 33-1, accepts the analog input from each of the telephones 38-1 to 38-4, converts the analog input to digital data, and packages the data into ATM packets (POTS over ATM), with each connection having a unique VPI/VCI. A VPI is a Virtual Path Identifier and a VCI is a Virtual Channel Identifier. The ATM network is connection oriented which can be characterized by a VPI/VCI pair. The VPI is used to identify a virtual path that exists over a physical medium. The VCI is used to identify the virtual channel within a virtual path. This leads to the concept of a PVC(permanent virtual circuit) or a SVC (switched virtual circuit). A PVC is a connection between

two endpoints and is usually manually configured. It is a path from end-to-end established using VPI/VCI value pairs at each node along the way through the ATM network. A SVC is similar to a PVC, except it is setup or torn down on a demand basis using a signaling protocol.

The Ethernet data is also encapsulated into ATM cells with a unique VPI/VCI. The ATM cell stream is sent to an internal DSL Modem within the CPE unit to be modulated and delivered to the DSLAM 34.

From the DSLAM end, the DSL signal is received and demodulated by the CPE DSL Modem and delivered to VPI/VCI Detection processing. The ATM cell data with VPI/VCI matching that of the end user's telephone is then extracted and converted to analog POTS to be delivered to one of the telephones 38-1 to 38-4. The ATM cell data with VPI/VCI matching that of the end user's Ethernet connection is extracted and delivered to an internal Ethernet transceiver for delivery to the Ethernet port. The above process will be described in depth in connection with detailed description of the CPE and components below.

#### **DSLAM 34**

The DSLAM 34 demodulates the data from multiple DSL modems 33-1 to 33-n and concentrates the data onto, for example, an ATM backbone network for connection to the rest of the network. The DSLAM 34 provides back-haul services for packet, cell, and/or circuit based applications through concentration of the DSL lines onto ATM outputs to the ATM Switch 35.

#### **ATM Switch 35**

The CO ATM switch 35 is the backbone of the ATM network. The ATM switch performs various functions in the network, among them:

- Cell Transport
- Multiplexing and concentration
- Traffic Control
- ATM-layer management

Of particular interest in the System Domain 30, the ATM switch provides for the cell routing and buffering in connection to the DSLAM 34, Service Control Processor 37 and the Internet Gateway 36, and T1 circuit emulation support in connection with the Public Switch 40 that handles voice telephone calls.

One exemplary embodiment of an ATM switch may be a Newbridge MainStreetXpress 36170 Multiservices Switch (Newbridge 36170). This switch has DSLAM capability and individual DSO control capability via T1 Circuit Emulation cards. It is possible that neither the

DSLAM capability nor the DSO control will be integrated into the switch, but rather in discrete components.

Another exemplary embodiment of an ATM switch may be a Lucent Access Concentrator with optional T1 cards added. The concentrator functions as the routing mechanism to aggregate and disperse data to various destinations within and outside the NCS, under the control of the SCP 37.

In one embodiment of the invention, optional T1 cards are installed in the ATM switch that allow ATM cells to be placed onto a DSO timeslot as part of a T1 connection. There may be, for example, 6 T1 ports per card. In this case, the T1 card is connected to the Telco's switching network.

Another exemplary interface from the ATM switch 37 to the telco switch 40 may be via a GR-303 interface. The GR-303 interface defines an Interface Group (IG) which may contain from 2 to 28 DS1 connections. Channels 12 and 24 are used for control information on the first DS1 connection. If there is a second DS1 in the IG, it contains redundant control information on channels 12 and 24 also. These channels are only used on the first 2 DS1 connections. All other DS1 connections utilize all 24 channels for voice traffic.

For the 1 or 2 DS1s that contain control information, channel 12 is used as the Timeslot Management Channel (TMC) and channel 24 is used as the Embedded Operations Channel (EOC).

A block diagram of this interface is shown in Fig. 12.

#### **Internet Gateway 36**

The Internet Gateway 36 provides for the ATM signaling support of internetworking of IP over ATM in connection to and from the ATM switch 35. An exemplary embodiment of an Internet Gateway may be a Cisco 7200 Series High Performance Router. This router has the capacity for four ATM ports and 32 10Base-T Ethernet ports. Other models may also be chosen based on capacity and performance.

#### **Service Control Processor (SCP) 37**

The SCP 37 provides for address translation, demand assignment and call management functions similar to an Internet Domain Name Service (DNS) server. The SCP 37 would be available for other functions as well, such as downloading code to the CPE and bandwidth and call management (e.g., busy) functions as well as other service provisioning and setup tasks.

One exemplary embodiment of a SCP is a Performance 500 series PC, on sale from Gateway, populated with one or two Fore Systems ForeRunnerHE 155 ATM Server Adapters. The ForeRunnerHE 155 supplies the SCP with full duplex OC-3 connectivity to the Newbridge 36170 which may serve as an ATM switch. One of the ATM adapters will serve as the dedicated command link to the Newbridge 36170. If the Newbridge 36170 supports a Proxy Server type of control and if such a control channel requires a separate port, then the second ATM adapter would be installed in the SCP. This ATM adapter would then serve as the data path for a PVC between the SCP and the Newbridge 36170.

Fig. 13 shows another exemplary embodiment of an SCP in distributed environment. The decision units (CPUs) 1305-1 to 1305-n will consist of one or more off-the-shelf industrial packaged (rackmount) computers running an operating system capable of performing the functional tasks. Note that different CPUs may be running different operating systems if various software requires a specific platform in order to operate. The desired platform should be scalable so that additional processing power can be added with a minimum amount of re-work or design changes.

As shown in the figure, all components are connected via an internal private LAN 1310 (e.g., Ethernet) for ease of communication and connectivity. The data storage array 1311 comprises of network attached storage units that are connected to the LAN 1310 (e.g., an ethernet) for ease of communication and connectivity. The data storage array 1311 comprises of network attached storage units that are connected to the LAN 1310. This will allow direct access to the data through the remote access device in the event access is required for status and/or troubleshooting.

The ATM interface block 1320 is the interface that transforms ATM cells into data that can be processed by the CPUs. In an alternative, an ATM NIC on one or more CPUs could be used for this purpose.

Control signals to and from the DSLAM and ATM switch are also required for NCS operation. They may be in-band or out-of-band signals.

Figure 14 is a block diagram that shows how voice channels are allocated between a CPE and the Telco, thru a SCP, in accordance with the present invention. A SCP - CPE logical connection is used to control the data paths between the CPE unit, the ISP and Telco. All data paths between the CPEs and the NCS will have PVCs already established. This will allow 24-hour access to the ISP and a channel

that will allow the exchange of control information between the SCP and the CPE.

The voice channels will also be PVCs (each with a variable rate) but only between the CPE and the ATM switch, as shown in Fig. 14. The SCP will have to establish a connection in real time between the Telco and the voice channel, through, for example, an ATM switch, as calls are requested. This is due to the limited amount number of DSOs between the NCS and the Telco. Each voice data path will require a DSO in order to connect up to the Telco. For economic reasons, the number of DSOs will be less than the total number of possible voice channels. Because of this, the SCP will have to dynamically allocate DSOs as required to the CPE voice channels. The SCP will be responsible for handling and establishing all voice connections for both incoming and outgoing calls to the NCS and its CPEs.

As shown in the Fig. 14, the total number of voice channels from CPEs (XN) is much greater than the number of available DSO slots (M) leaving the NCS. Therefore, the SCP must allocate the DSO channels dynamically as calls are connected to the CPEs. The SCP has a command channel to the ATM switch that it uses to issue commands to the ATM switch so that it can route the CPE voice channels to the T1 interface and ultimately to the Telco.

When a voice call data path to the Telco is established, the ATM switch is commanded to route the appropriate VPI/VCI to one of the DSO slots on one of the T1 cards. The tracking of which DSOs and T1 ports are available is maintained by the SCP. All ATM cell conversions are handled by the T1 cards.

The SCP controls all aspects of the ATM switch including initial set-up. The control of the ATM switch can be achieved in one or more of the following ways:

1. through the RS232 port using terminal emulation. This is out-of-band signaling.
2. through the Ethernet port using Telnet. This is out-of-band signaling.
3. allocating a PVC between the SCP and the ATM switch. The SCP runs an SNMP client and controls the switch by using in-band signaling; and/or
4. through the use of the API on the ATM switch (the PacketStar Signaling Gateway). This is in-band signaling.

Use of in-band signaling will probably provide the fastest response from the ATM switch but with increased complexity of the

interface. In addition, in-band signaling can use already established ATM data paths. This will be especially useful in the event that the SCP and the ATM switch are not co-located.

Note that out-of-band signaling can utilize the ATM data paths already in existence and provide pseudo in-band signaling. For example, through the use of Local Area Network Emulation (LANE) provided by ATM.

The SCP may communicate with the IP Internet gateway 36 through the RS232 port or by using an SNMP client via TCP/IP.

The T1 interface to the Telco switch 40 transforms the ATM cells into data that is put onto a DSO channel and transmitted through a T1 line to the Telco switch. The reverse happens when data is received at the NCS.

The T1 interface consists of line cards that are installed as options on the ATM switch. The SCP will route data to an individual DSO slot on a T1 card by issuing a command to the ATM switch.

#### **Detailed CPE unit Implementation**

An exemplary detailed embodiment of a CPE unit 33-n is shown in circuit block diagram form in Fig. 4. The embodiment of the CPE unit architecture may comprise of, for example, an Altera 10K100A 240 pin FPGA 45, two Alcatel MTK-40131 SH POTS Evaluation Boards 44-1 and 44-2, an Alcatel MTK-20140 DSL Modem Evaluation Package 46, and several peripheral devices such as memory SRAM 413 and Ethernet physical layer processor 420. Each of these major CPE unit components is described in detail in the paragraphs that follow. These components may be used to test and evaluate, as well as implement the principles of the present invention.

One skilled in the art will readily recognize that the components selected and described herein for implementing a CPE unit, for example, CPE 33-1, in accordance with the present invention are exemplary only. They are chosen mainly for the purpose of being able to implement the principles of the present invention expediently, using readily available off-the-shelf products. Other circuits or integrated circuits performing the same functions of these components as described herein may also be used, as will be apparent to one skilled in the art.

#### **Telephone Interface - Alcatel MTK-40131 SH POTS Evaluation Boards 44-1 and 44-2**

The Alcatel MTK-40131 Evaluation Boards 44-1 and 44-2 are used as interface boards to POT telephones 38-1 to 38-4 (shown in Fig. 1) via

RJ11 connections 41-1 to 41-4. These boards comprising functions such as, for example, Analog to Digital Conversion (A/D), Digital to Analog Conversion (D/A), and line interface control, necessary to connect analog telephone sets or other analog terminals into digital communications systems.

As shown in Fig. 4, each of the MTK-40131 Evaluation Boards contains three devices: two Short-Haul Line Interface Circuits (SH LIC) 42-n that provide the signal and power interface to the analog lines (one per line) and a DSP based CODEC (CODSP) 43-n that provides all signal processing functions for up to two independent two-wire phone lines.

The digital interface to the Alcatel MTK-40131 Evaluation Boards is the General Circuit Interface (GCI). Ring signaling, on-hook signaling, off-hook signaling, and alarm detection are all provided to the Altera FPGA 44 by the Alcatel MTK-40131 Evaluation Boards, via the GCI interface 1 or 2.

The two Alcatel MTK-40131 Evaluation Boards 44-1 and 44-2 provide analog POTS to digital interfacing for up to four analog telephones 38-1 to 38-4. Of course, more or less of these Evaluation Boards may be used and the rest of the circuit scaled up or down accordingly, to provide capacity to accommodate more or less than 4 analog telephones.

For downlink data (i.e., towards an end-user telephone), the CODSP 43-1 or 43-2 routes incoming GCI data to one of the MTC-30132 SH LIC units for D/A conversion and delivery to one of the analog phone ports 41-1 to 41-4. For uplink data (i.e., towards the network), the MTC-30132 SH LIC units 42-1 to 42-4 convert incoming phone port analog data to digital and deliver the digital data to the CODSP 43-1 and 43-2. The CODSPs then deliver the digital data to the FPGA 45 via the GCI interfaces, as pointed out previously.

#### **ADSL Modem - Alcatel MTK-20140 Rate Adaptive ADSL Modem Evaluation Package 46**

The Alcatel MTK-20140 chipset 46, as shown in Fig. 4, consists of an Analog Front-End (MTC-20144) 47 and a digital chip (MTC-20146) 48. MTC-20146 integrates three components and or functions:

- DMT (Discrete Multi-Tone) modem
- Dedicated ADSL Transceiver Controller with associated firmware
- ATM Framer

The MTK-20140 46 therefore provides all the active functions required to implement a complete ATM-based, rate adaptive DMT ADSL modem.

The data interface between the MTK-20140 46 and the Altera  
FPGA 45 is implemented as an ATM Utopia interface 410. The  
MTK-20140 also provides a POTS splitter 414 with connectivity to  
two RJ11 ports 49-1 and 49-2. RJ11 49-1 may be used for  
connection to the telephone wire pair from the central office, for  
example, while the other jack, 49-2, may be used for connection to  
an analog POTS telephone.

#### **ATM Processing - Altera 10K100A 240 pin Field Programmable Gate Array (FPGA) 45**

The Altera FPGA 45 (which will now be referred to simply as FPGA)  
has been programmed to provide general ATM processing for the DSL  
service. In particular, functions of the FPGA 45 comprise ATM cell  
stream processing, control signal processing, GCI interfacing to the  
MTK-40131 SH POTS units 44-1 and 44-2, UTOPIA interface and  
control to the MTK-20140 ADSL modem 46, and traffic control and  
shaping for the CPE unit in general.

#### **DSO / ATM Interworking Functions (IWF's)**

On the FPGA uplink path (CPE to Network), 56 kbps data will be  
packaged into ATM cells as part of the ATM Cell Encapsulation  
function provided by FPGA 45. The 56 kbps data is the data to be  
carried on a DSO channel.

The issue of how much bandwidth is required to transfer a single  
DSO seems harmless and straightforward until one reads the ATM  
Forum's af-vtoa-0078 (*Circuit Emulation Service Interoperability  
Specification*) and af-vtoa-0085 (*Specifications of (DBCES)  
Dynamic Bandwidth Utilization – In 64 kbps Time Slot Trunking  
Over ATM – Using CES*) specifications.

According to af-vtoa-0078, a single DSO could require the entire T1  
bandwidth if it is the only active channel in the T1. This may be  
fine for a switching fabric, but it is not fine for the ADSL uplink  
path. The af-vtoa-0085 specification allows for not sending idle  
channels, but bandwidth is still consumed by ATM pointer  
overhead. Both specifications allow (actually necessitate) the  
usage of IWF's to aid in efficient bandwidth management.

In the DSL system in accordance with the present invention, the  
DSO data must be packed into ATM cells as efficiently as possible  
(in terms of bandwidth usage). There are IWF's on both the FPGA  
uplink and downlink paths as well as in the network. On the CPE



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uplink path, the IWF packs the 56 kbps data (in octets) into an ATM cell payload, and waits until the payload is full before transmitting the cell. This introduces a small delay (about 6 msec) in the transmitted data.

5 On the CPE downlink path, the IWF extracts the payload data and delivers it to the MTK-40131 SH POTS unit at a rate of 56 kbps. The network IWF's are analogous to the CPE IWF's. The network IWF's may be contained in an addressable box between the ATM switch and a T1 CES card. Alternatively, there may exist a T1 CES  
10 card that performs the network IWF as needed for the DSO data.

### FPGA Downlink Path

The ADSL modem 46 sends data to the FPGA 45 via the UTOPIA interface 410. The UTOPIA data enters the Traffic Control and  
15 Shaping function 411 in FPGA, which routes the data to the internal ATM Cell Bus 412. Each of the nodes on the internal ATM Cell Bus 412 performs VPI/VCI detection and filtering on the incoming ATM cells, as shown for examples, as blocks 420-1 to 420-6.

20 If a VPI/VCI detection function detects a VPI/VCI match, the processing node passes the ATM cell data to the ATM cell payload extraction function, shown as blocks 421-1 to 421-6.

For the telephone ports, the ATM cell payload extraction function sends the extracted payload data to the MTK-40131 SH POTS unit via the GCI bus protocol. For the Control Process path 422, the ATM cell  
25 payload extraction function sends the extracted payload data to the Control Process function 423.

Among other things, the Control Process function 423 provides VPI/VCI assignment for the other CPE ports. For the Ethernet path, the ATM cell payload extraction function 421-6 sends the extracted  
30 payload data to the 10Base-T Physical Interface Chip (ML2653) 420.

The downlink payload extraction process for all of the ports contains an IWF (as described before) that gathers the DSO 64 kbps data from the ATM cells, and sends the data to the MTK-40131 SH POTS unit 44-1 or 44-2 at 56kbps, for further conversion to analog voice. A  
35 block diagram for performing this downlink operation within FPGA 45 is shown in Fig. 5.

In particular, the downlink Interworking Function 50 comprises a counter 51, FIFO 52 and Converter and Interface 53, whose function will be described in connection with tables 1-8 below.

As described before, an uplink ATM Cell Encapsulation Interworking Function is also needed. This is shown in Fig. 6 of the present invention.

As shown in Fig. 6, the telephone port data (via the GCI interface) and the Ethernet data 10Base-T Physical Interface Chip enter their respective ATM Cell Encapsulation functions at 60. The resulting ATM cells are sent to the VPI/VCI Generation function 420, which inserts VPI/VCI data into the cell and delivers the ATM cell to the Traffic Control / Shaping function 411. This data is then sent to the MTK-20140 ADSL modem unit 46 for transmission. The uplink ATM Cell Encapsulation function 60 for all of the ports contains an IWF that packs the DSO data into full ATM cell payloads and delivers the data to the ADSL modem.

The CPE FPGA 45 formats an AAL1 ATM cell for the telephone ports in the manner described by tables 1 through 8 using counter 61, F1F0 62, and converter 63. The Structured Data Transfer (SDT) pointer in this system always has a value of 0x00. The sequence count field that is contained in the tables refers to the sequence count field in the SN field defined in Section 2.4.2.1 of ITU-T I.363.1. The sequence count field counter 61 is a 3 bit field that starts with a value of 0 and increments in each successive ATM cell to a maximum of 7 and then wraps to a value of 0 (modulo 8). When the SN counter is 0, the SDT pointer is present in the ATM cell. This ATM cell stuffing format is known as 1x56 with CAS. The ABCD bits (CAS bits) are placed in the ABCD order, left justified ("A" is the MSB, "D" is bit 4). The lower nibble is stuffed with 0. The CAS bits are stolen (bit robbing) from the LSB of the DSO frames 6, 12, 18, and 24 of the Extended Superframe (ESF), composed of 24 DS1 frames plus formatting.

#### Sequence

##### Count 0

Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU Header	1
7	SDT Pointer	1
31	24 DSO Samples	24
32	CAS bits	1
53	21 DSO Samples	21

Table 1 Sequence Count Field = 0

#### Sequence

##### Count 1

Total Bytes	Data Type	# bytes
5	ATM Cell Header	5

		16	
6	AAL1 SAR PDU	1	
	Header		
9	3 DSO samples	3	
10	CAS bits	1	
34	24 DSO samples	24	
35	CAS bits	1	
53	18 DSO samples	18	

Table 2 Sequence Count Field = 1

## Sequence

## Count 2

Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU	1
	Header	
12	6 DSO samples	6
13	CAS bits	1
37	24 DSO samples	24
38	CAS bits	1
53	15 DSO samples	15

Table 3 Sequence Count Field = 2

5

## Sequence

## Count 3

Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU	1
	Header	
15	9 DSO samples	9
16	CAS bits	1
40	24 DSO samples	24
41	CAS bits	1
53	12 DSO samples	12

Table 4 Sequence Count Field = 3

## Sequence

## Count 4

Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU	1
	Header	
18	12 DSO samples	12
19	CAS bits	1
43	24 DSO samples	24
44	CAS bits	1
53	9 DSO samples	9

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Table 5 Sequence Count Field = 4

Sequence Count 5		
Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU Header	1
21	15 DS0 samples	15
22	CAS bits	1
46	24 DS0 samples	24
47	CAS bits	1
53	6 DS0 samples	6

Table 6 Sequence Count Field = 5

Sequence Count 6		
Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU Header	1
24	18 DS0 samples	18
25	CAS bits	1
49	24 DS0 samples	24
50	CAS bits	1
53	3 DS0 samples	3

Table 7 Sequence Count Field = 6

Sequence Count 7		
Total Bytes	Data Type	# bytes
5	ATM Cell Header	5
6	AAL1 SAR PDU Header	1
27	21 DS0 samples	21
28	CAS bits	1
52	24 DS0 samples	24
53	CAS bits	1

Table 8 Sequence Count Field = 7

5

The FS6370 EEPROM Programmable Clock Generation IC 425 provides clocking for the FPGA and several peripherals. A 16 MHz input 426 to the FS6370 is used because this frequency facilitates generation of many other needed system clock frequencies. The 20 MHz clock signal is used for Ethernet processing and the 512 kHz clock signal is used to support the GCI interface. In another implementation, a master system clock may replace this 512 kHz clock signal.

### **SRAM 413**

A SRAM 413 will be used by the FPGA 45 as scratchpad RAM in case there is not enough memory inside the FPGA to handle all of the FIFO's and packetizing RAM needed for the ATM and Ethernet processing functions.

### **BIT Blaster Port 430**

FPGA 45 is SRAM-based and requires configuration after power-up. The BIT Blaster port is an on-board header that is used to configure the FPGA 45 during the prototype development only. The EPC1 serial EPROM must not be installed when using the BIT Blaster port. The on-board EPC1 serial EPROM 435 will configure the FPGA during normal operations.

### **Ethernet Interface 420**

The 28-pin ML2653, 10BASE-T Physical Interface Chip 420, is a complete physical interface for twisted pair and AUI Ethernet applications. It combines a 10BASE-T MAU, Manchester Encoder/Decoder, and Twisted Pair Interface filters in one monolithic IC. The ML2653 420 can automatically select between an AUI and twisted pair interface based on Link Pulses. The ML2653 conforms to IEEE 802.3I—1990 (10Base-T), and together with the header formation in the FPGA, it will accept and provide Ethernet packets pursuant to the G.802 standard for packet structure.

### **Data Path Logical Flow**

Fig. 15 shows the logical flow of the data paths from the CPE through the NCS and back again (bi-directional). Also shown are the data formats used between the different functional blocks.

The voice data paths may be connected to a Telco's network via T1 links as described before. A T1 consists of 24 voice channels called DSO's as shown in. Hence each voice data path uses one of 24 DSO channels of the T1 connection.

The voice data paths are variable bit rate (VBR) while the command and PC paths are unspecified bit rate (UBR) as shown, for example, in Fig. 17.

Fig. 16 is a block diagram of the data paths between the CPEs and the NCS.

Note that the Command and PC data paths are UBR while all of the voice paths are specified as VBR and are assigned a unique VPI/VCI for each path. Refer to the Voice Traffic Management section for a description of how the NCS manages the data path allocation for the voice channels. The voice data is formatted using as-vtoa-0078 on AAL1.

The Command channel is a non-blocking channel to the NCS (and ultimately to the SCP) so that a CPE is always in contact with the NCS. Bandwidth will have to be allocated so that the maximum number of CPEs have sufficient bandwidth to allow communication with the NCS.

The PC data path is also a non-blocking channel but could drop to a minimum bandwidth in the event of network congestion. This non-blocking channel requires that sufficient bandwidth will have to be allocated accordingly.

As ATM cells move through the NCS, the cells can have their ATM addresses (VPIs/VCIs) translated. In addition, the ATM cells that carry voice data can be blocked at the DSLAM. Fig. 16 is an example of the data paths for ATM cells from two CPEs.

In this example, there are 2 active voice channels; Voice 1 of CPE 1 and Voice 3 of CPE 2. The other voice channels are "blocked" at the DSLAM since they are not carrying any data. Refer to the Voice Traffic Management paragraph for a description of how this occurs.

### **Operational Call Flow Sequences**

The following sections describe how each major components of the present invention operate in response to different call flow scenarios.

Fig. 7 shows the operational scenario diagram for placing or initiating a call. The sequence enumeration is further described in the paragraphs that follow.

1. One of the 4 phone lines on the CPE goes off hook.
2. The corresponding LIU signals an off hook indication to the CPE processing unit.

3. The CPE processing unit recognizes the change of state of the off hook.

4. The CPE processing unit formats an ATM cell containing a unique CPE identifier, the off hook indication, and the telephone port ID on the CPE.

5. The CPE issues the ATM cell to the network through the DSL interface.

6. The DSLAM passes the ATM cell to the ATM switch, which routes it to the SCP.

7. The Service Control Processor (SCP) recognizes the request from the CPE.

8. The SCP determines the availability of a DSO slot connecting to the public network..

8 (a). The SCP determines that no DSO is available.

8 (b). The SCP generates user feedback. In this case, that feedback is nothing, to emulate the telephone network when a connection to a switch is not available.

9. The SCP creates a VP/VC path connecting the available DSO slot to the DSLAM port of the CPE.

10. The SCP checks it's database and retrieves the phone number corresponding to the CPE port.

11. The SCP informs the T1 line card of the calling (source) phone number.

12. The SCP creates an ATM cell, which contains the VP/VC of the established connection and a confirmation message of the connection.

13. The cell created by the SCP is transmitted through the ATM switch to the port of the ATM switch connected to the DSLAM and on to the CPE. This cell VP/VC terminates at the CPE processing unit.

14. The CPE processing unit receives the ATM cell.

15. The CPE processing unit configures the VP/VC generation/detection to the directed VP/VC.

16. The ATM VP/VC channel begins carrying the DS0 payload between the LIU of the CPE all the way to the T1 line emulation cards in the switch.

17. Call progress, tone generation, and tone detection proceeds between the telephone unit and the public telephone network.

Fig. 8 shows the scenario diagram for the call in process operational sequence.

Fig. 9 shows the scenario diagram for the CPE call disconnect operational initiated by an CPE. The sequence enumeration is further described in the paragraphs that follow.

1. One of the 4 phone lines on the CPE goes on hook during an established call in Progress.
2. The corresponding LIU signals an on hook indication to the CPE processing unit.
3. The CPE processing unit recognizes the change of state of the on hook.
4. The CPE processing unit formats an ATM cell containing a unique CPE identifier, the on hook indication, and the telephone port ID on the CPE.
5. The CPE issues the ATM cell to the network through the DSL interface.
6. The DSLAM passes the ATM cell to the ATM switch, which routes it to the SCP.
7. The SCP recognizes the request from the CPE.
8. The SCP informs the T1 line emulation of the call disconnect.
9. The SCP tears down VP/VC path connecting the available DS0 slot to the DSLAM port of the CPE.
10. The SCP creates an ATM cell that acknowledges the call tear down.
11. The cell created by the SCP is transmitted through the ATM switch to the port of the ATM switch connected to the DSLAM and on to the CPE. This cell VP/VC terminates at the CPE processing unit.
12. The CPE processing unit receives the ATM cell.



13. The CPE processing unit removes the VP/VC generation/detection to the disconnected port.

Fig. 10 shows the operational scenario diagram for incoming call. The sequence enumeration is further described in the paragraphs that follow.

1. The SCP initially is monitoring the status of the T1 interface connected to the public telephone network.
2. A DS0 channel indicates an incoming call.
3. The SCP determines the destination telephone number from the T1 line emulation cards.
4. The SCP looks up the CPE port ID in a database relating phone numbers and Port Ids
5. The SCP creates a VP/VC connection to the DSLAM of the CPE containing the targeted phone port.
6. The SCP creates an ATM cell containing information on the incoming call, the targeted port, and the established VP/VC. This cell is transmitted to the CPE processing unit.
7. The CPE processing unit receives the cell and configures the targeted port with the VP/VC.
8. The CPE rings the phone at the appropriate RJ11 port.
9. At this point, the connection is set up from the T1 line emulator to the telephone port of the CPE.

Fig. 11 shows the operational scenario diagram a call disconnect initiated by the telco switch or NCS. The sequence enumeration is further described in the paragraphs that follow.

1. The SCP initially is monitoring the status of the T1 interface connected to the public telephone network.
2. A DS0 channel indicates a termination of an existing call.
3. The CPE device user goes on-hook.
4. The corresponding LIU signals an on-hook condition to the CPE processing unit.
5. The CPE processing unit recognizes the change of state of the on-hook.

6. The CPE processing unit formats an ATM cell containing a unique CPE identifier, the on hook indication, and the telephone port ID on the CPE.
7. The CPE issues the ATM cell to the network through the DSL interface. The DSLAM passes the ATM cell to the ATM switch, which routes it to the SCP.
8. The SCP recognizes the request from the CPE.
9. The SCP informs the T1 line emulation of the call disconnect.
10. The SCP tears down VP/VC path connecting the available DSO slot to the DSLAM port of the CPE.
11. The SCP creates an ATM cell that acknowledges the call tear down.
12. The cell created by the SCP is transmitted through the ATM switch to the port of the ATM switch connected to the DSLAM and on to the CPE. This cell VP/VC terminates at the CPE processing unit.
13. The CPE processing unit receives the ATM cell and removes the VP/VC generation/detection to the disconnected port.

#### **Voice Traffic Management**

Voice traffic management refers to the control of data that carries voice information throughout the NCS and how it is managed when there is less system capacity than the theoretical maximum amount of data that could pass through the system. For example, if there are 100 telephone lines connected to the NCS but only 10 DSO lines connected to the Telco, the DSO lines must be allocated dynamically as connections are required for each telephone connection desired.

Statistics will be used to allocate an economical amount of physical data bandwidth at various points in the system. Since this number will be less than the theoretical maximum amount of data, there are different points in the system where data blocking could occur (i.e., a voice connection cannot be completed). The SCP will be responsible to manage the allocation of data channels as requested. And, in the event that a request cannot be fulfilled, the SCP is responsible to gracefully exit from that condition and inform the requesting entity of the non-compliance.

## 24

Fig. 18 is an example of the data paths for voice traffic through the NCS. Note that each CPE shows a voice channel going to the NCS via a PVC.

As shown in the figure, there is a potential of a call being blocked at the DSLAM or at the Telco switch. The potential block at the DSLAM is determined by the size of the DSLAM-to-ATM switch data path. The potential block at the Telco switch is determined by the number of DSO lines.

Each voice channel PVC is established when the CPE is connected to the system. The PVC is specified as a Variable Bit Rate (VBR). This allows the DSLAM and ATM switch to be "over subscribed". That is, the ATM switch will allow all of the connections to be established since they are set up as VBR. The CPE will not transmit data on any voice channel until commanded to do so by the SCP.

In the example shown in Fig. 18, each DSLAM has only enough bandwidth to support a single voice channel to the ATM switch. However, it is possible that 3 channels could attempt to establish a connection. Prior to a connection, none of the CPEs are transmitting data on the voice channel, therefore the DSLAM is not passing data to the ATM switch. This allows the DSLAM-to-ATM switch to be completely unused in this example. Once a connection is established between a CPE (e.g., CPE 2) and the Telco, the SCP will command CPE 2 to start transmitting voice data on its data channel. Now the data path between DSLAM 1 and the ATM switch is full. No other voice connections can be made until CPE 2's connection is removed. Even though a PVC has been established between each CPE and the ATM switch, the SCP has control over the allocation of bandwidth by commanding each CPE to transmit or not transmit voice data.

### **Non-Voice Traffic Management**

The data paths that do not carry voice data (i.e., the Command and PC data channels) are non-blocking and will have a minimum bandwidth available for each path. Traffic management is less of a concern for the SCPs in this case since the initial set-up of each data path will specify the path as UBR with a minimum data rate. The ATM devices will enforce this data rate as a function of the ATM protocol itself. As system throughput changes, the bandwidth for each channel will adjust accordingly based on the amount of data attempting to be passed through that channel.

This will require that each potential "choke point" in the data path must be implemented so that it will be able to handle the minimum

25

amount of required bandwidth for the number of CPEs connected to the NCS.

Fig. 19 is an example block diagram of where throughput can be reduced for a system consisting of 6 CPEs.

5

Each of the PVCs in the figure has a potential to be limited at each location as indicated. Therefore, each device in that data path must be able to accommodate the required amount of bandwidth (including transmission protocol overhead) for all non-blocking data channels connected to the system.

Claims

1. A method for providing a voice connection through a data network, comprising the steps of:

5 requesting a connection;

formatting the connection request into a data cell;

10 sending the data cell to a service control point;

determining in real time whether a connection to public switch network is available; and

15 if the connection to a public switch network is available, setting up the requested connection as a virtual circuit connection to the public switch network.

20 2. The method of claim 1 further comprising the step of setting up the requested connection through a data packet network.

3. The method of claim 2 wherein the data packet network is an asynchronous transfer mode network.

25 4. The method of claim 2 wherein requesting step is made through a permanent virtual circuit already established to the data packet network.

5. The method of claim 1 wherein the requested connection is setup as ATM adaptation layer type 1.

30 6. The method of claim 1 wherein the data cell is in asynchronous transfer mode format.

35 7. The method of claim 1 wherein the virtual circuit connection is a switched virtual circuit connection.

8. The method of claim 1 wherein the service control point processes the connection request in real time in response to the data cell.

40 9. The method of claim 1, further comprising the step of tearing down the virtual circuit connection when the requested connection is terminated.

10. A system for providing a data and/or voice service, comprising:

45 a device for detecting a connection request from an end user terminal and formatting the connection request into a data packet;

27

a service control point for receiving the data packet and for determining whether a connection to a public switch network is available; and

5 if the connection to the public switch network is available, the service control point setting up the requested connection as a virtual circuit connection to the public switch network.

10 11. The system of claim 10 wherein the requested connection is setup through a data packet network.

12. The system of claim 11 wherein the data packet network is an asynchronous transfer mode network.

15 13. The system of claim 11 wherein the device sends the data packet through a permanent virtual circuit already established to the data packet network.

20 14. The system of claim 10 wherein the requested connection is made through an ATM adaptation layer type 1 connection.

15. The system of claim 10 wherein the data packet is in asynchronous transfer mode format.

25 16. The system of claim 10 wherein the virtual circuit connection is a switched virtual circuit connection.

17. The system of claim 10 wherein the service control point processes the connection request in real time in response to the data packet.

30 18. The system of claim 10, wherein the service control point tears down the virtual circuit connection when the requested connection is terminated.

35 19. The method of providing an end-to-end voice connection for an end device, comprising the step of:

setting up a permanent virtual circuit connection from the end device to a data network;

40 setting up a switched virtual circuit connection through the data network; and

connecting the switched virtual circuit to a circuit switched connection in a voice network.

45 20. The method of claim 19 wherein the permanent virtual circuit connection is variable rate.

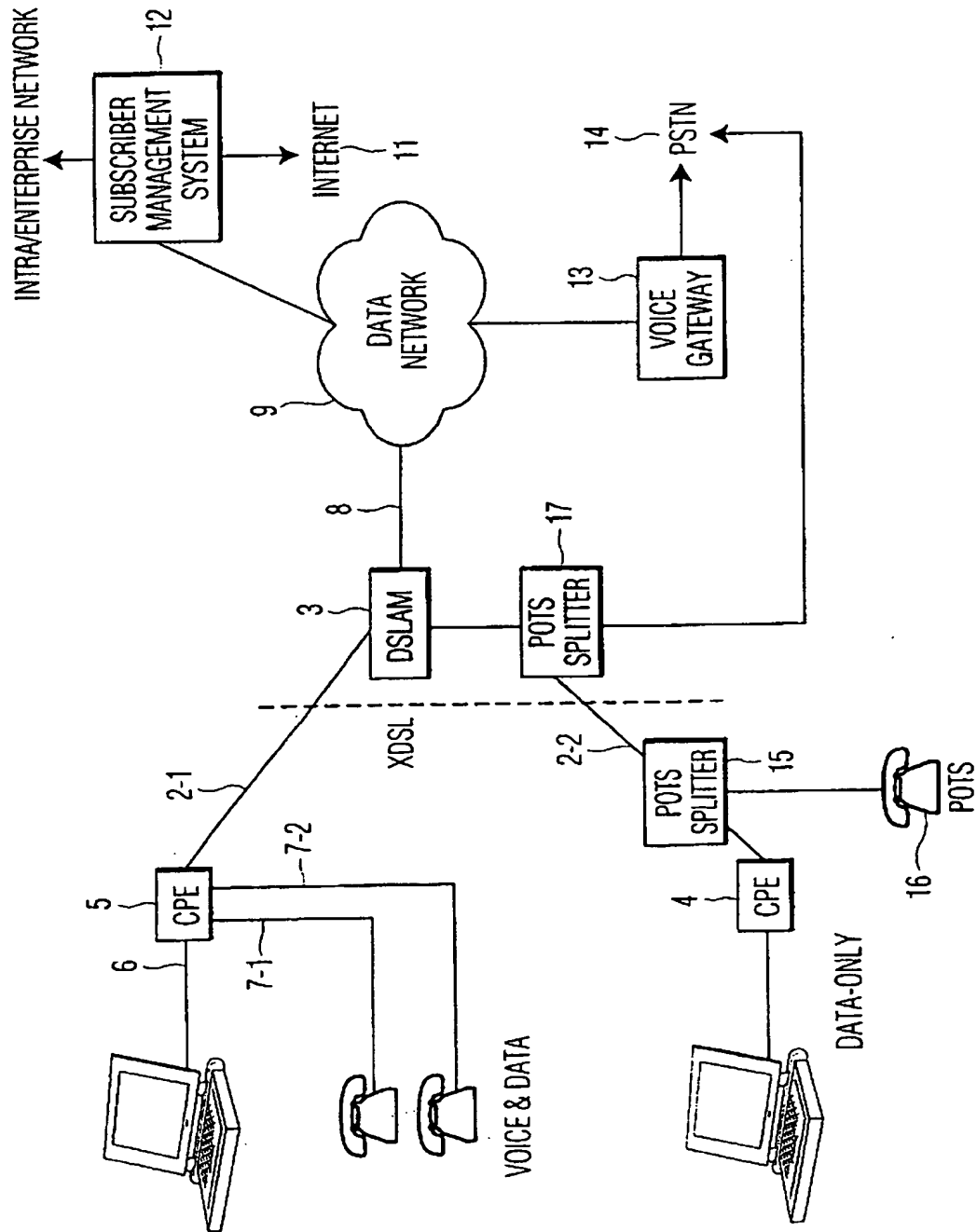


FIG. 1  
PRIOR ART

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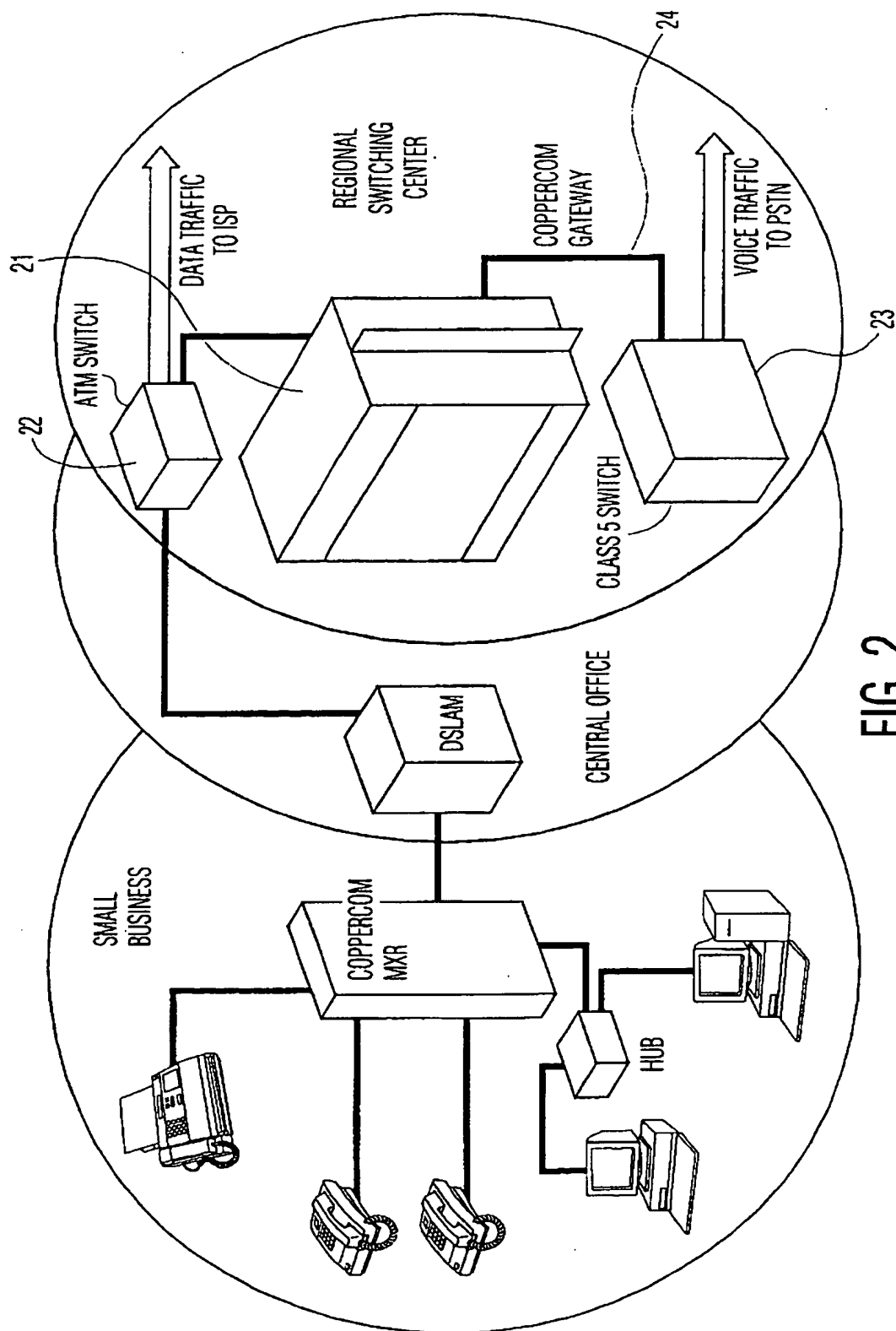


FIG. 2



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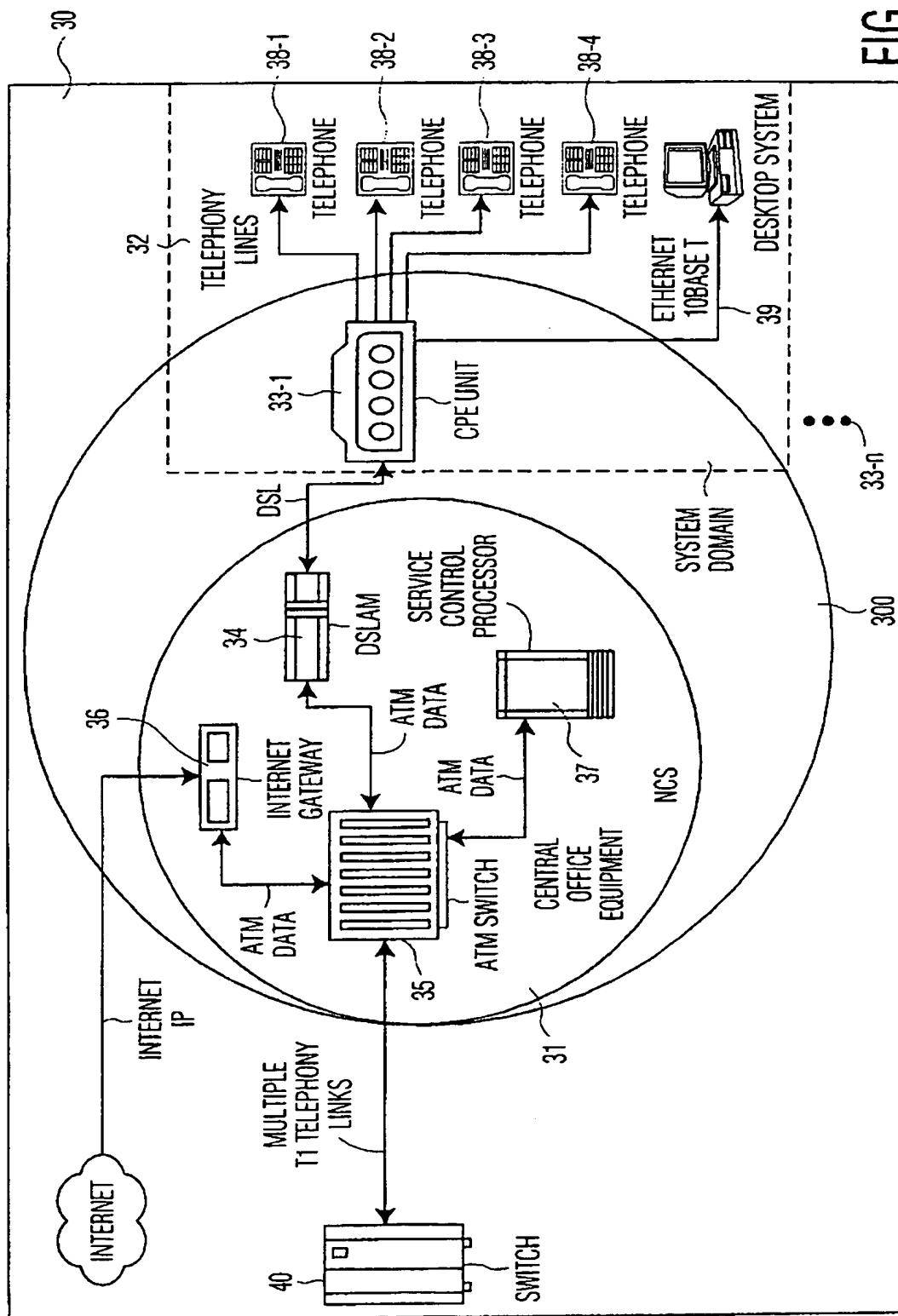
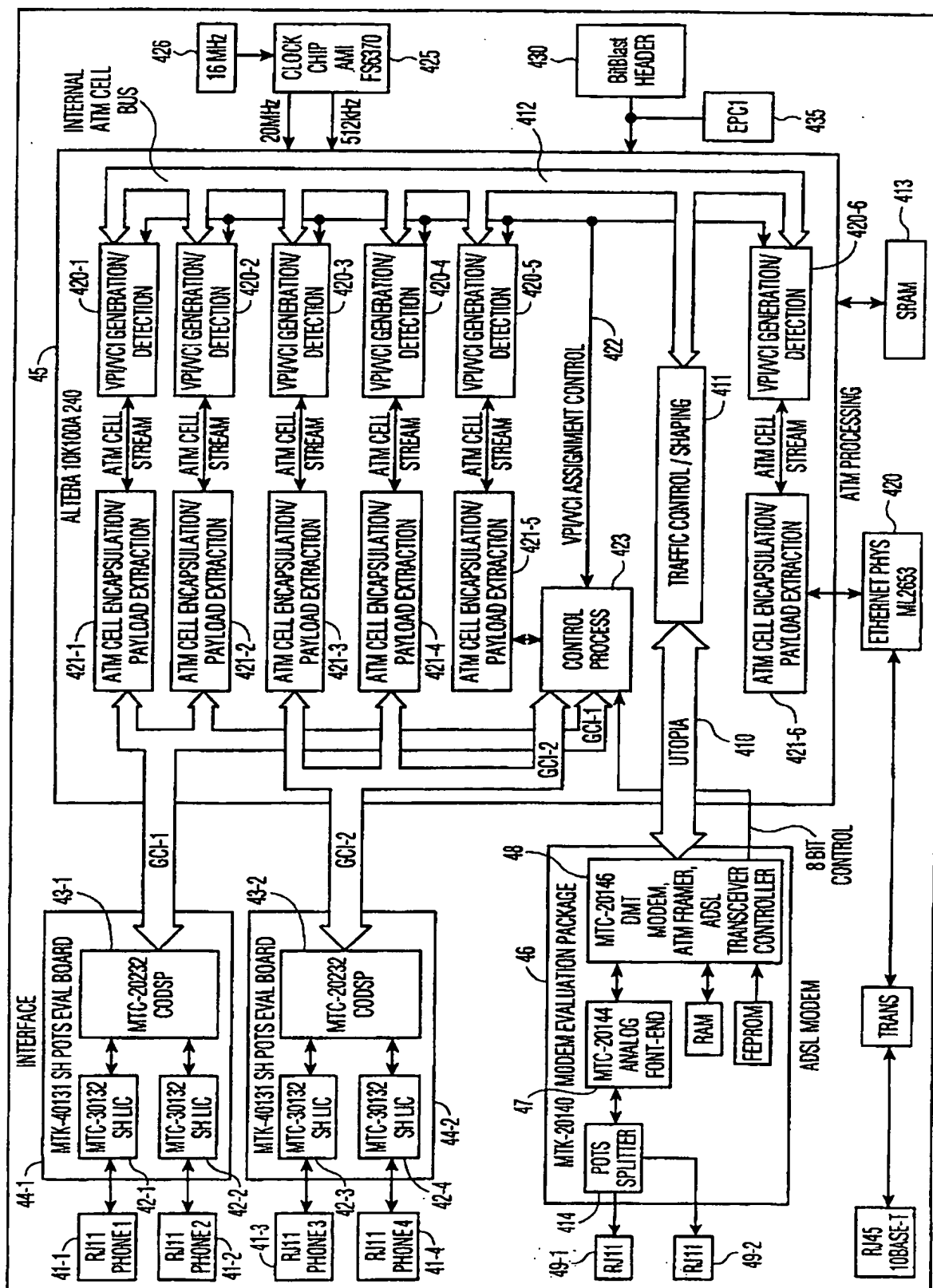


FIG. 3



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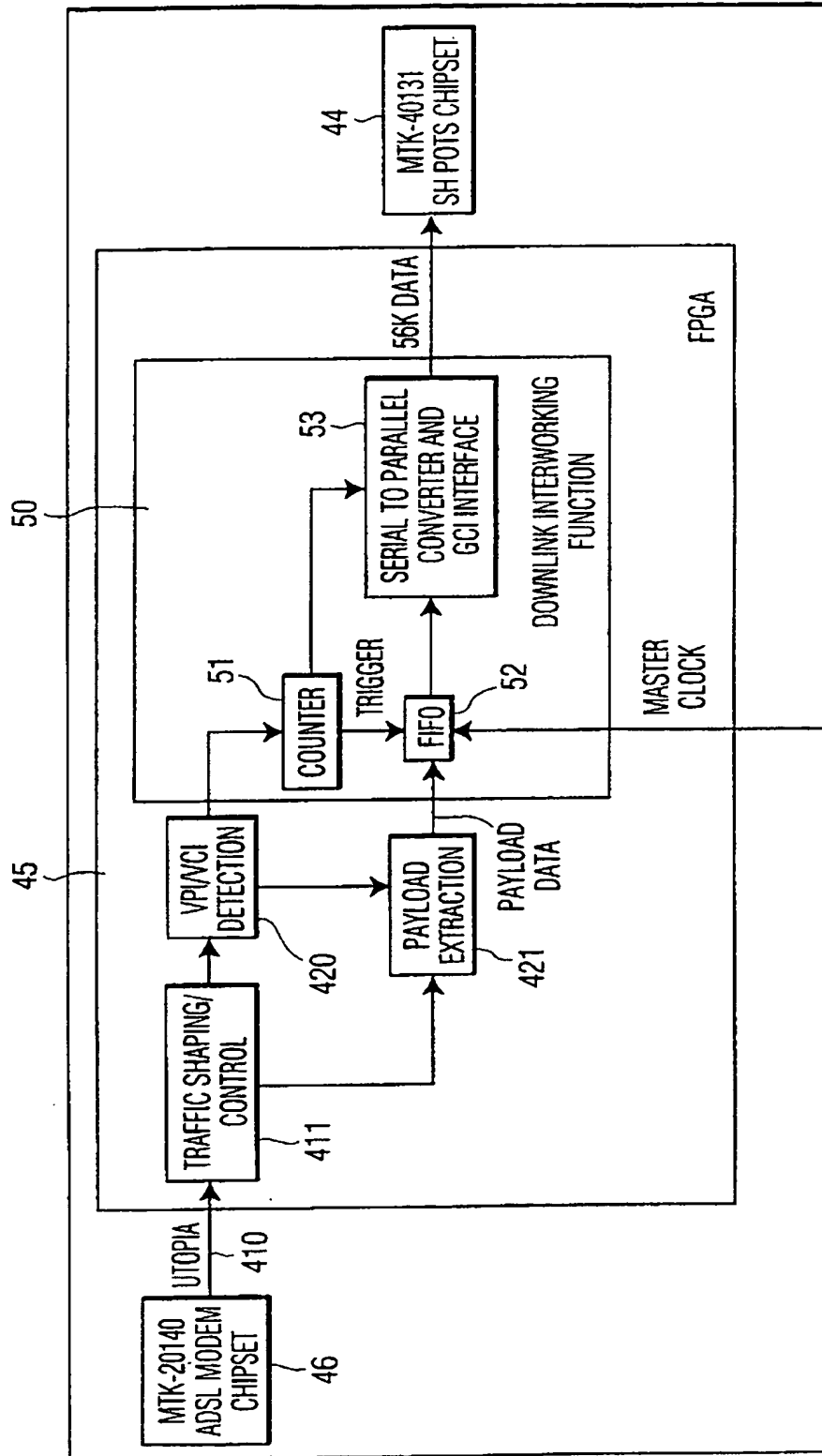


FIG. 5

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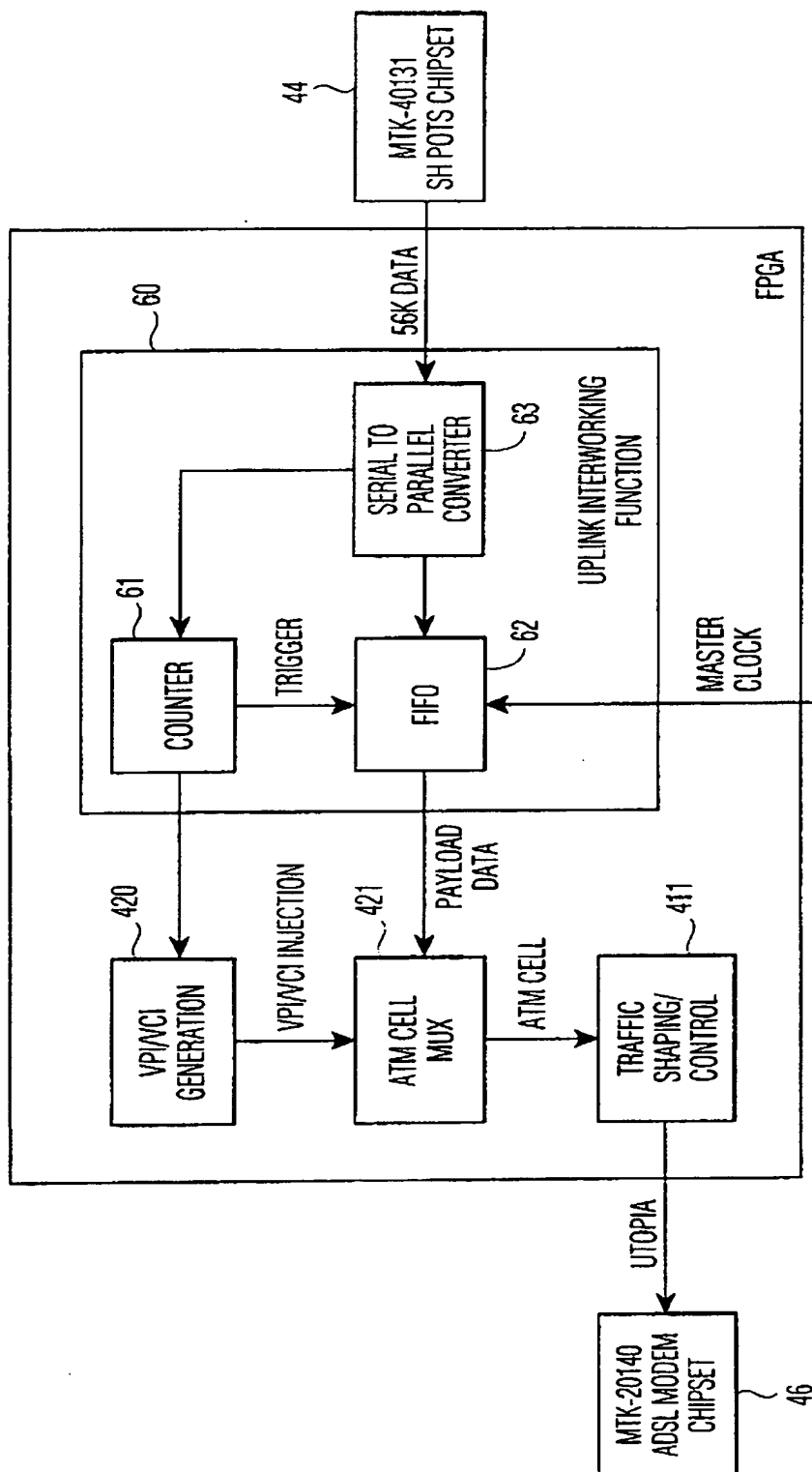
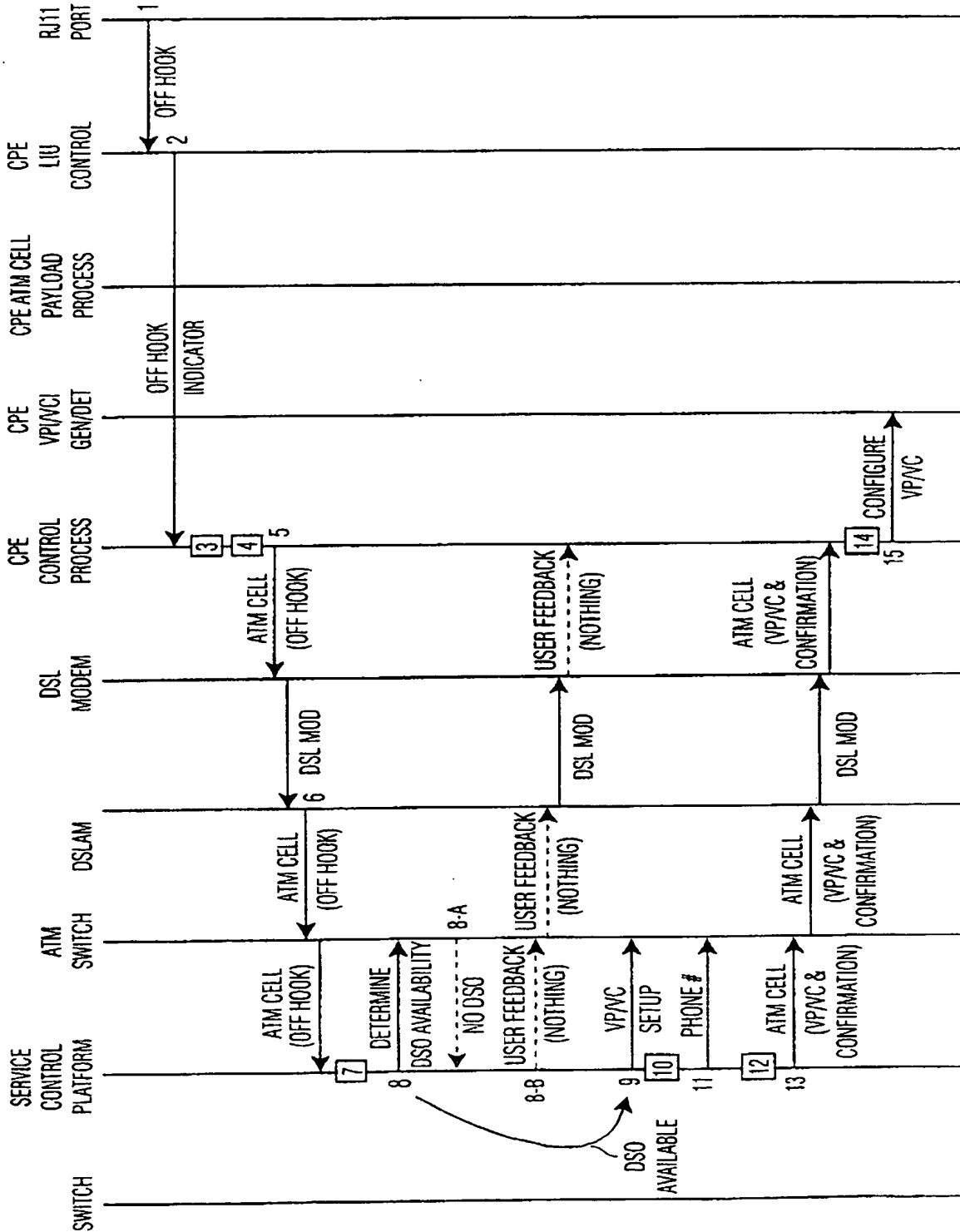


FIG. 6

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FIG. 7



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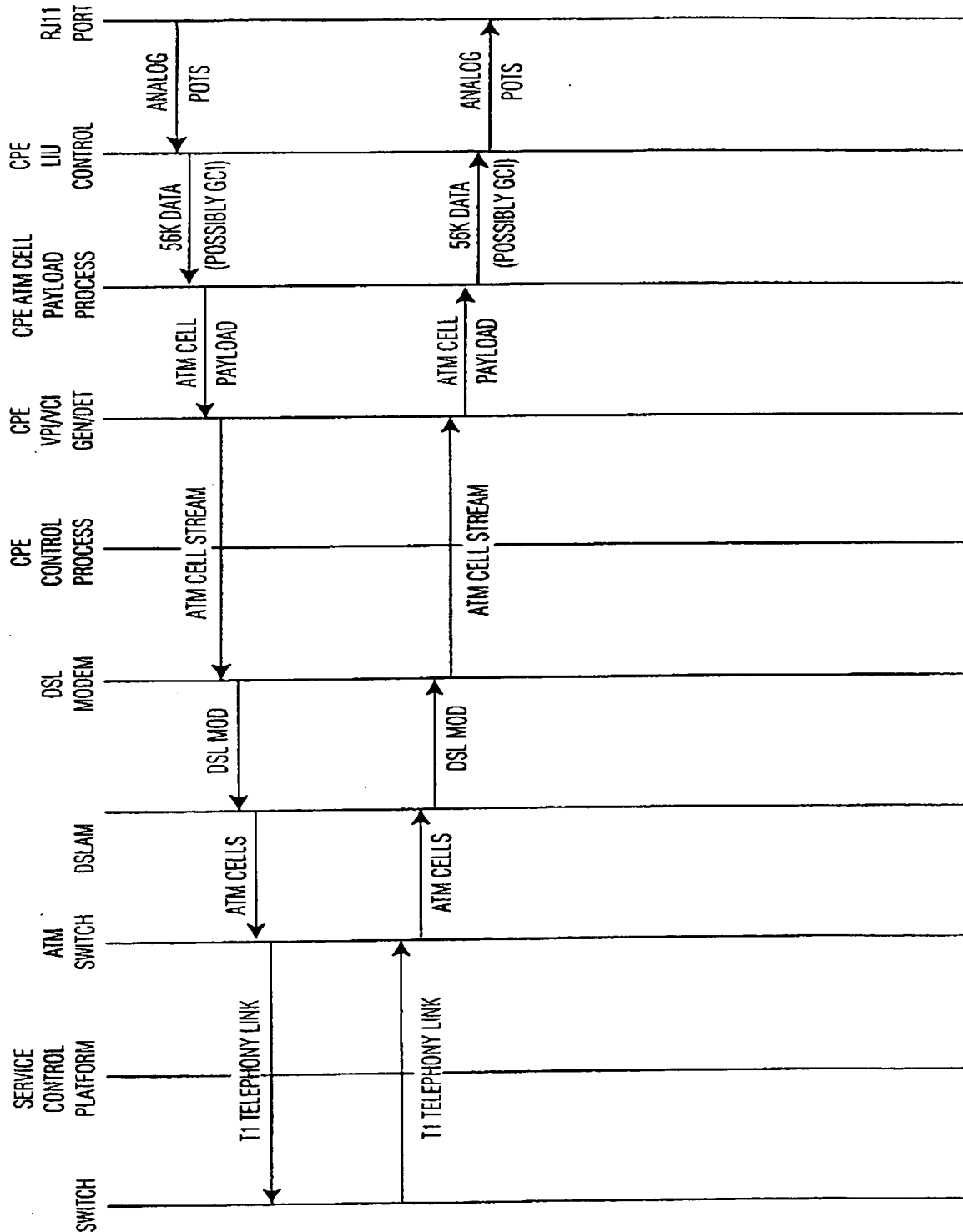


FIG. 8

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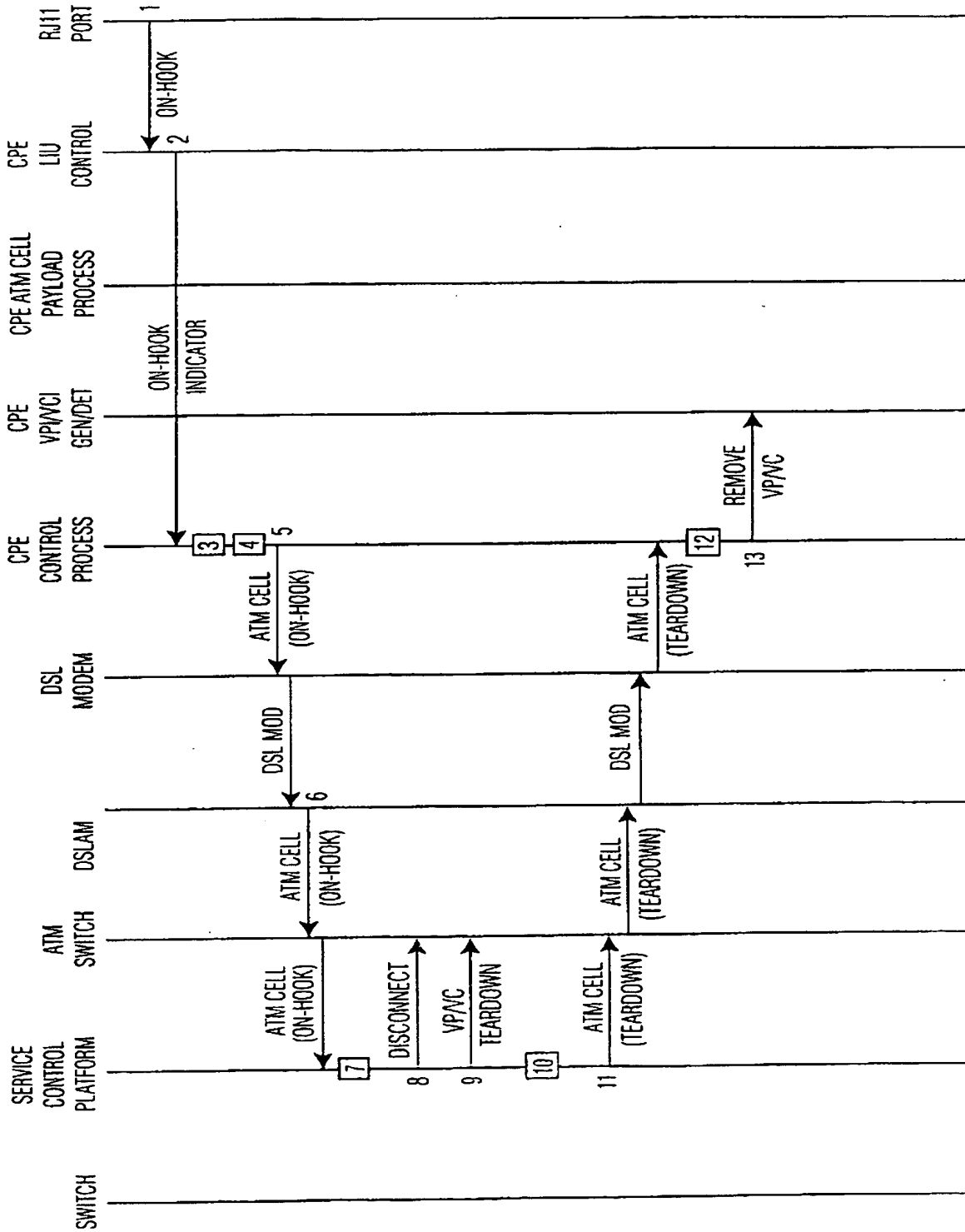
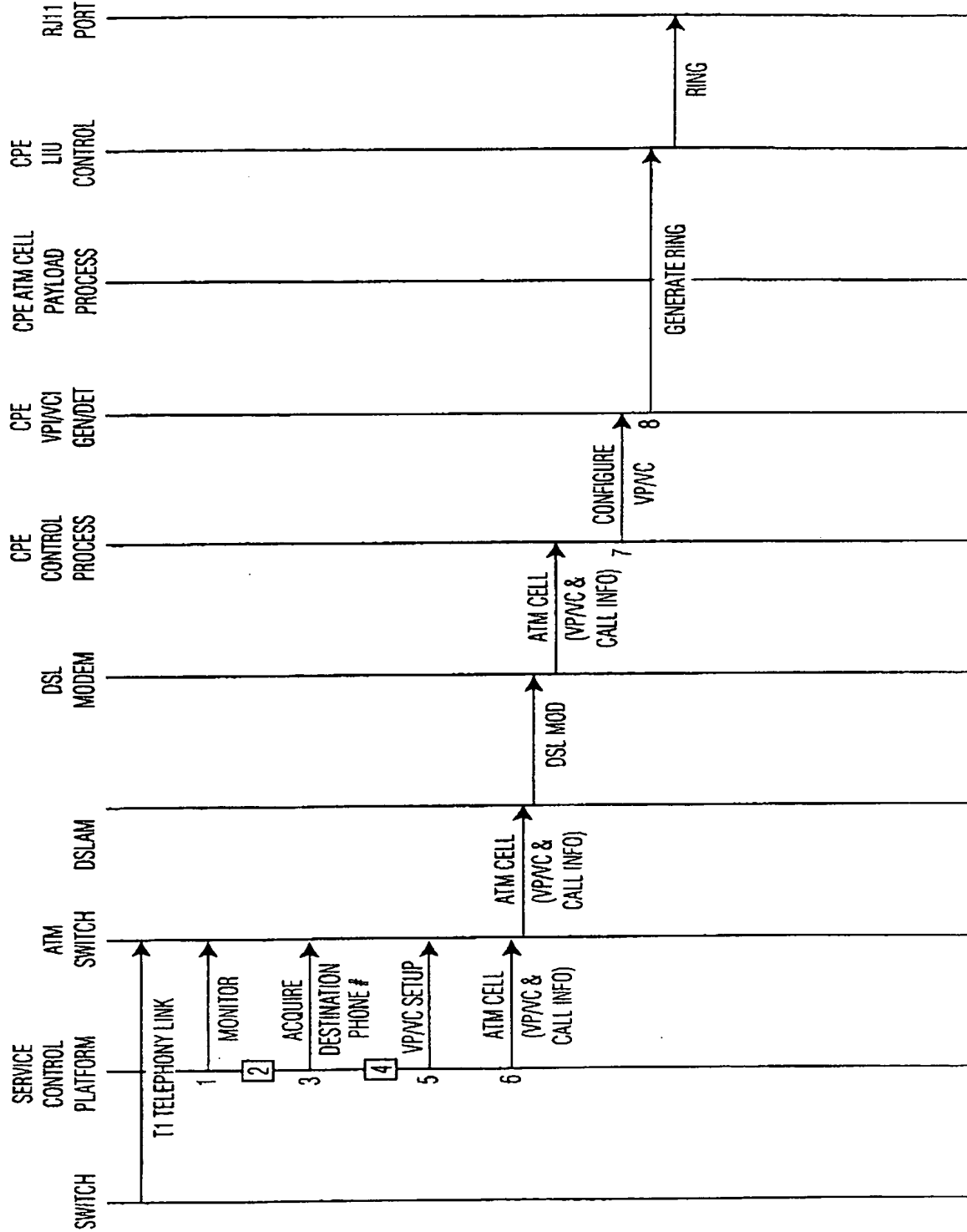


FIG. 9

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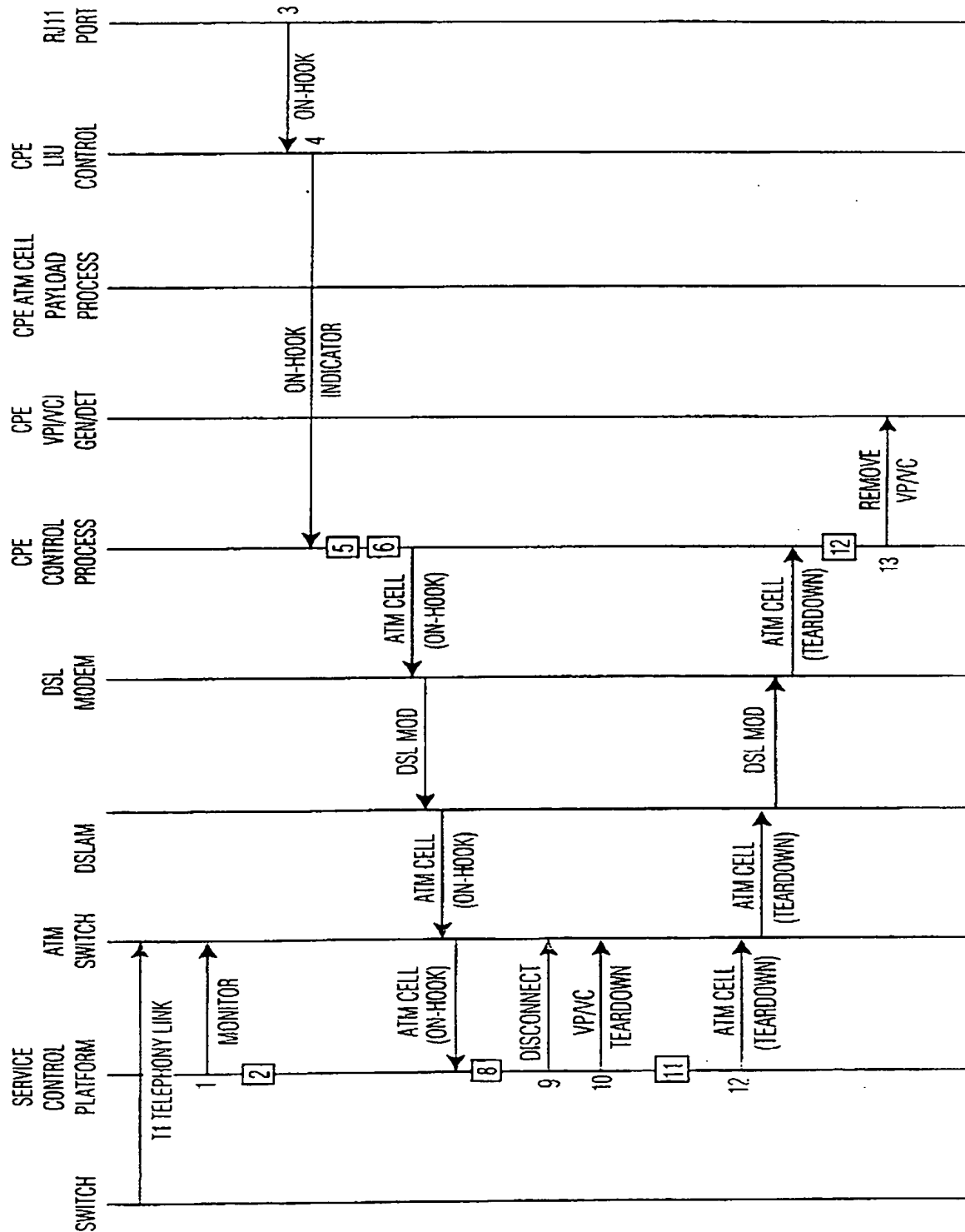
FIG. 10





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FIG. 11



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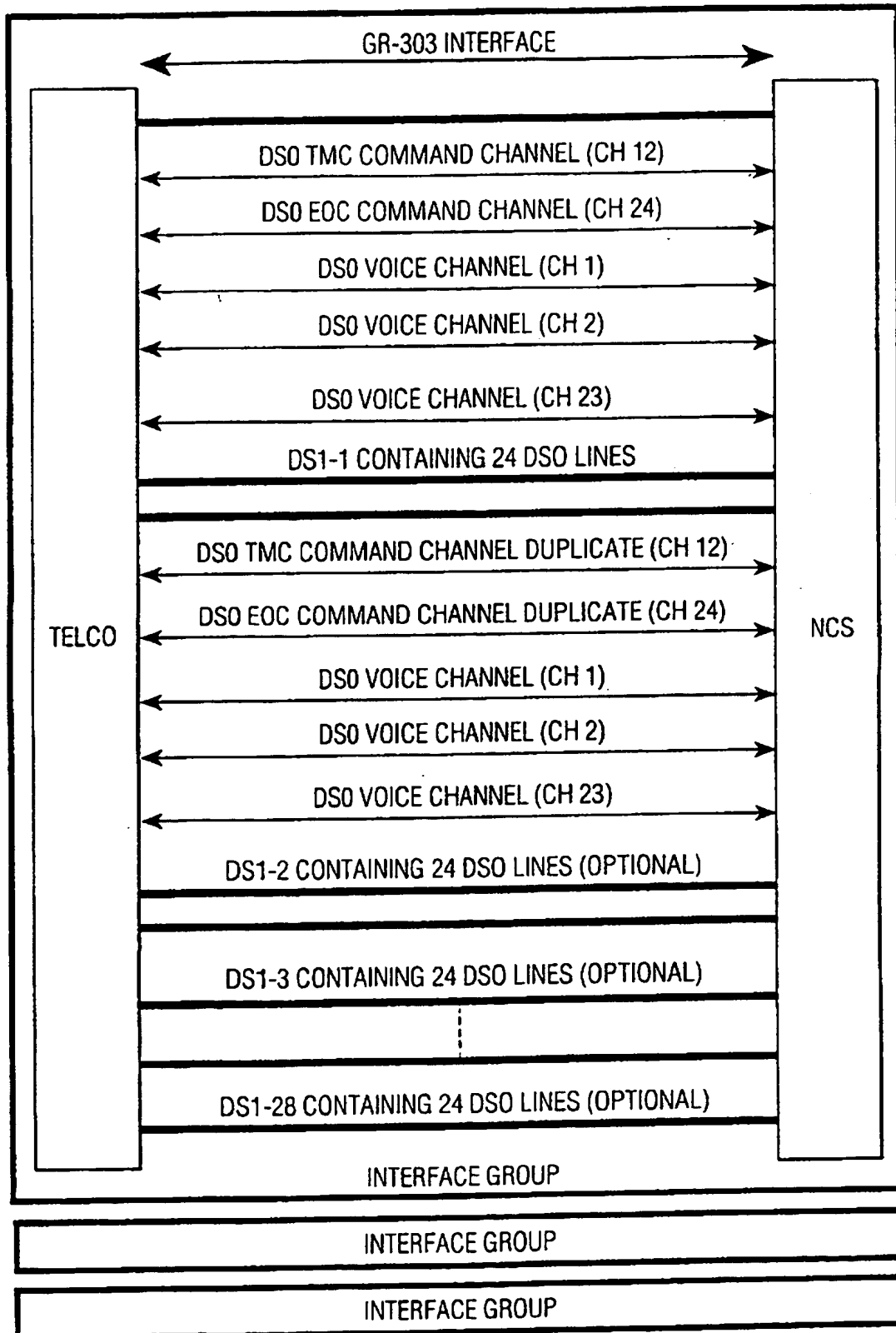


FIG. 12

SUBSTITUTE SHEET (RULE 26)

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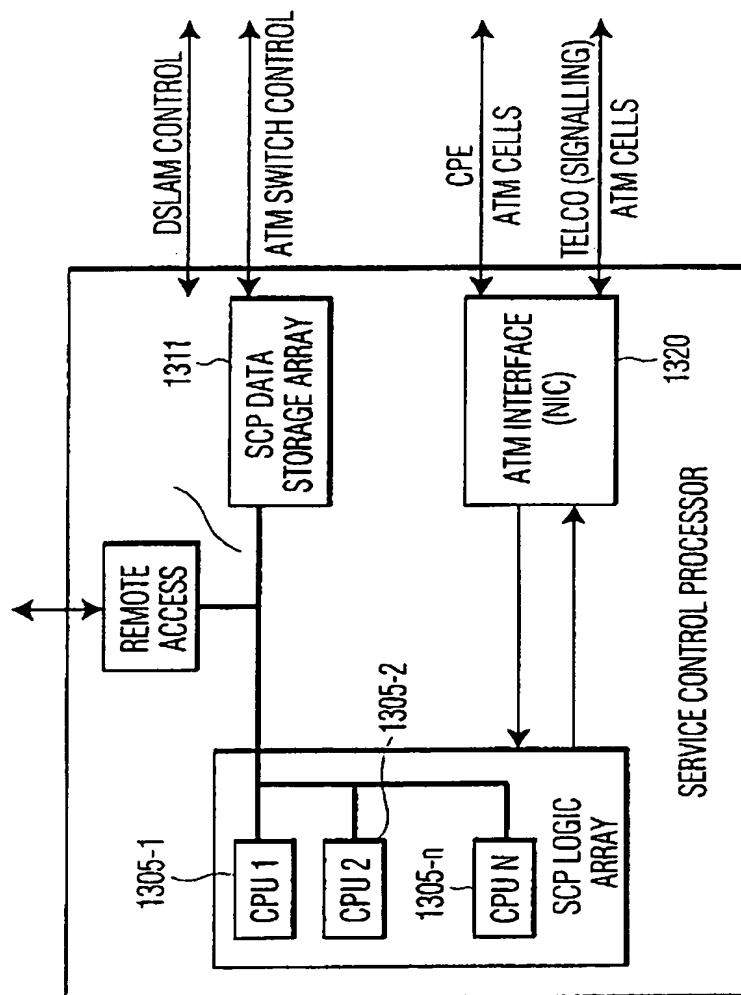


FIG. 13

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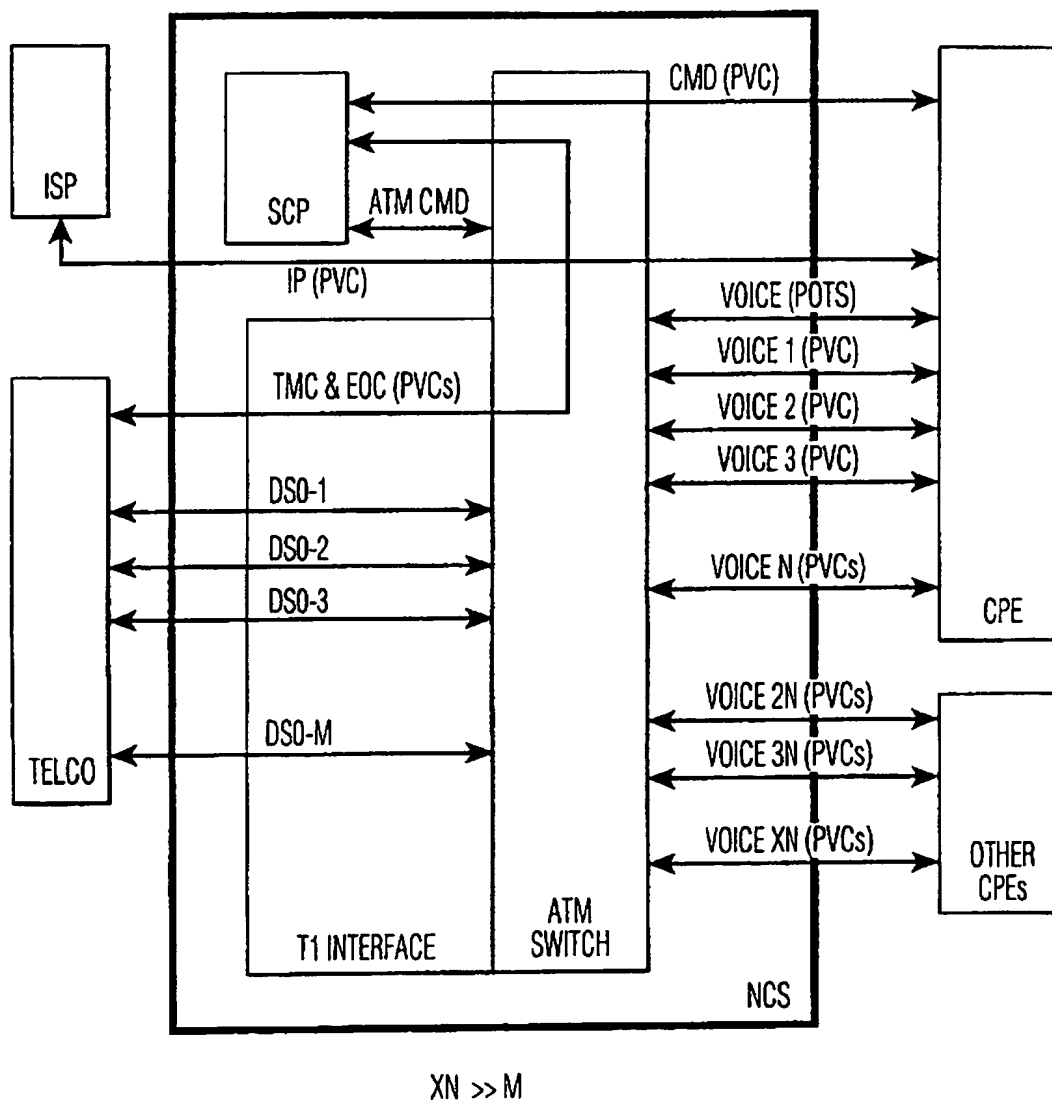


FIG. 14

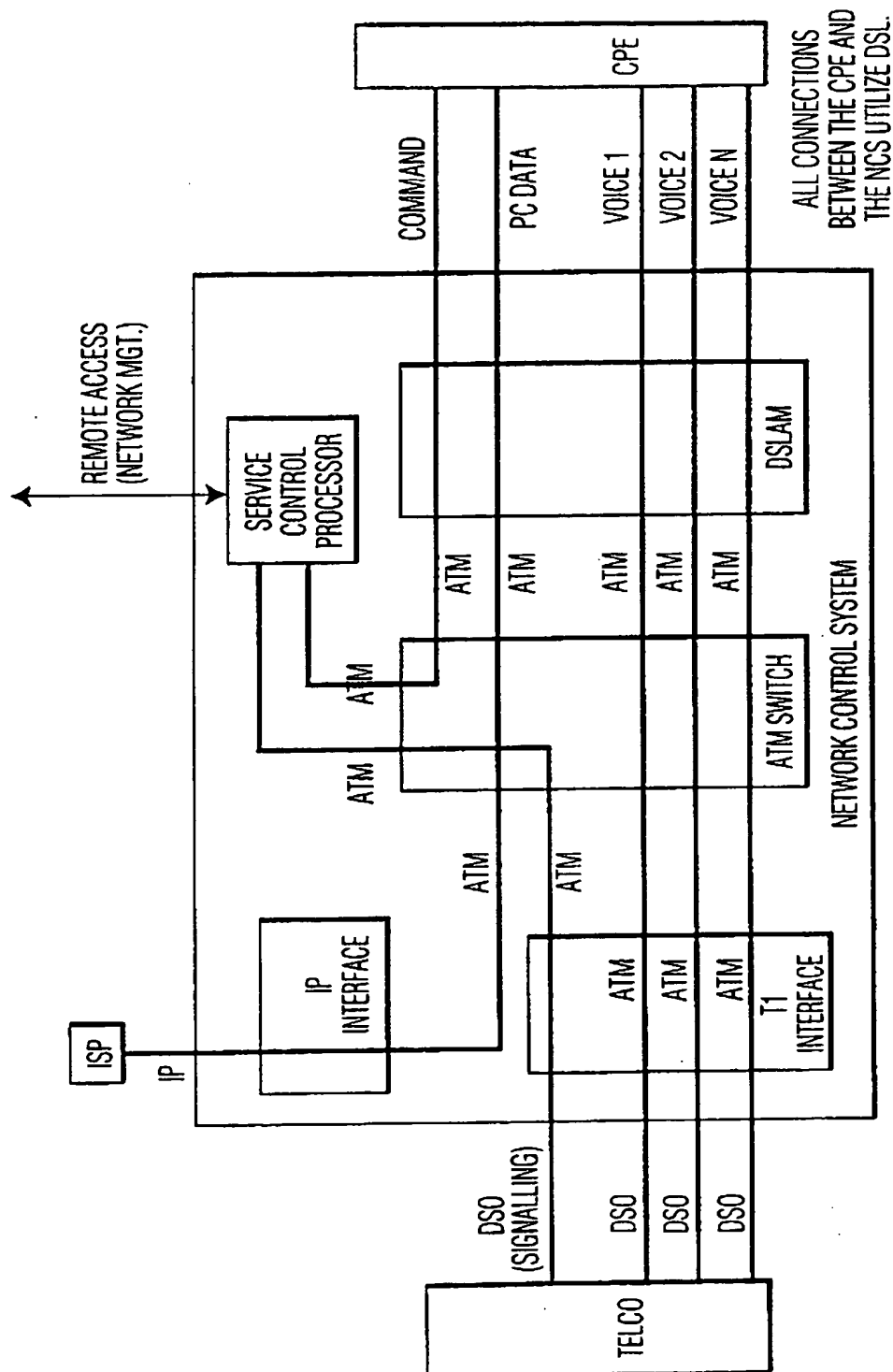


FIG. 15

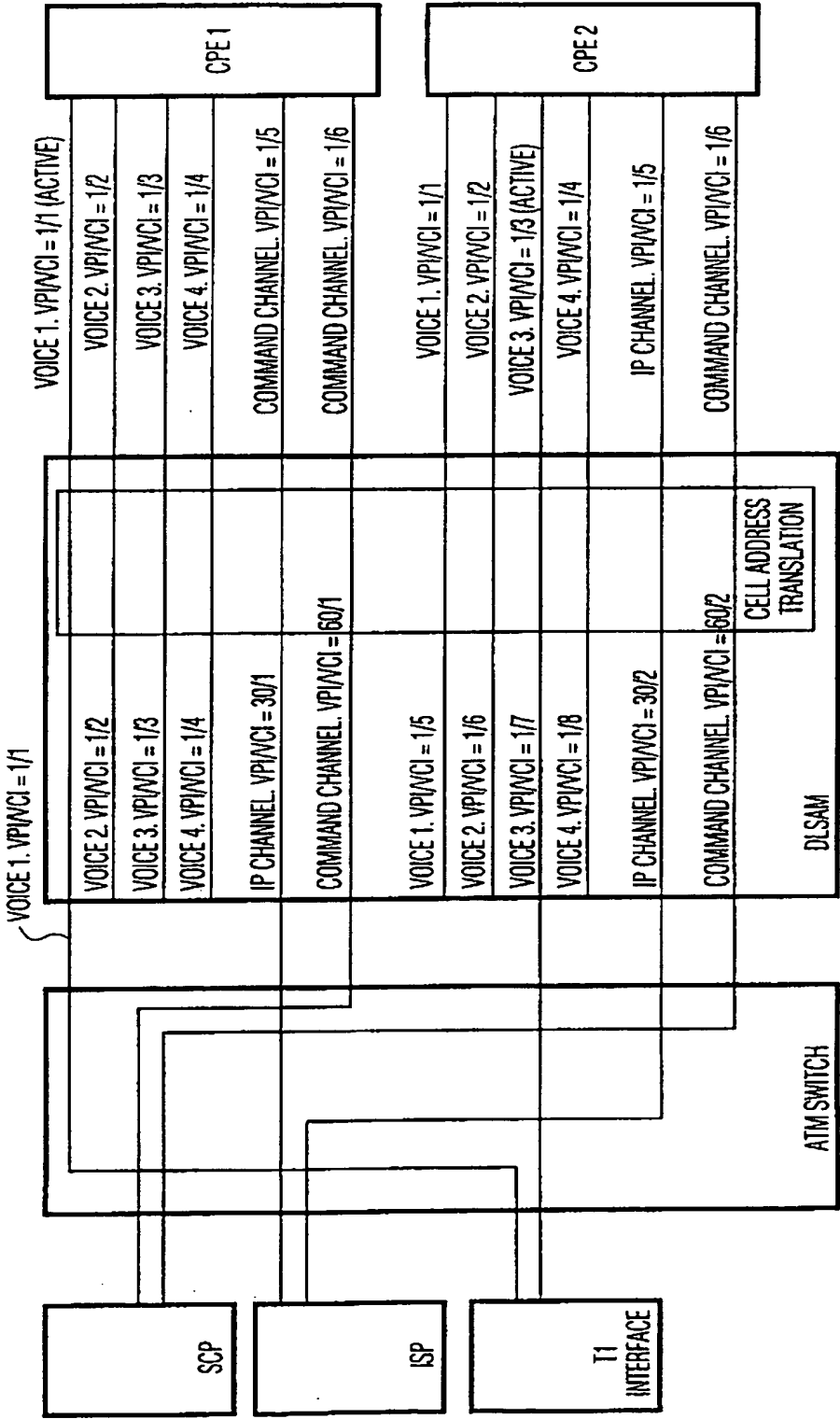


FIG. 16

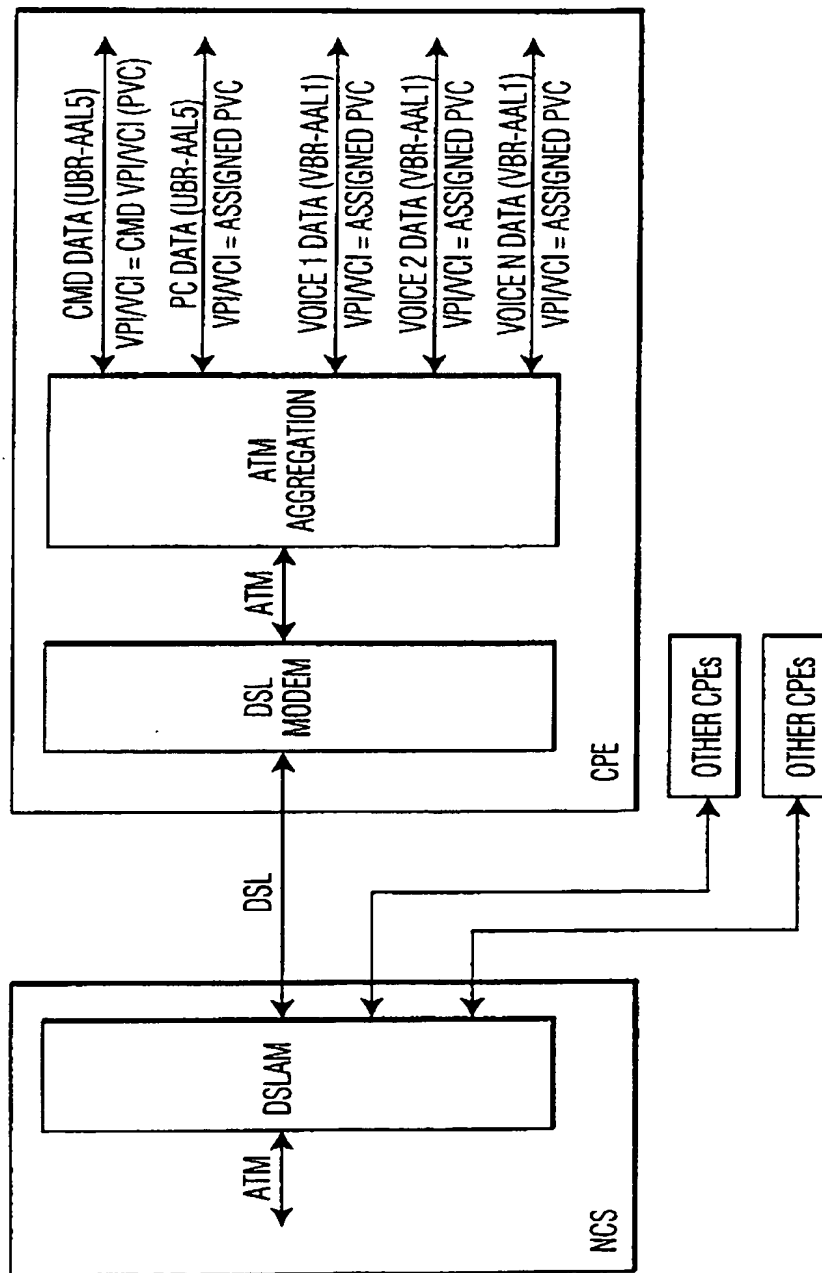


FIG. 17

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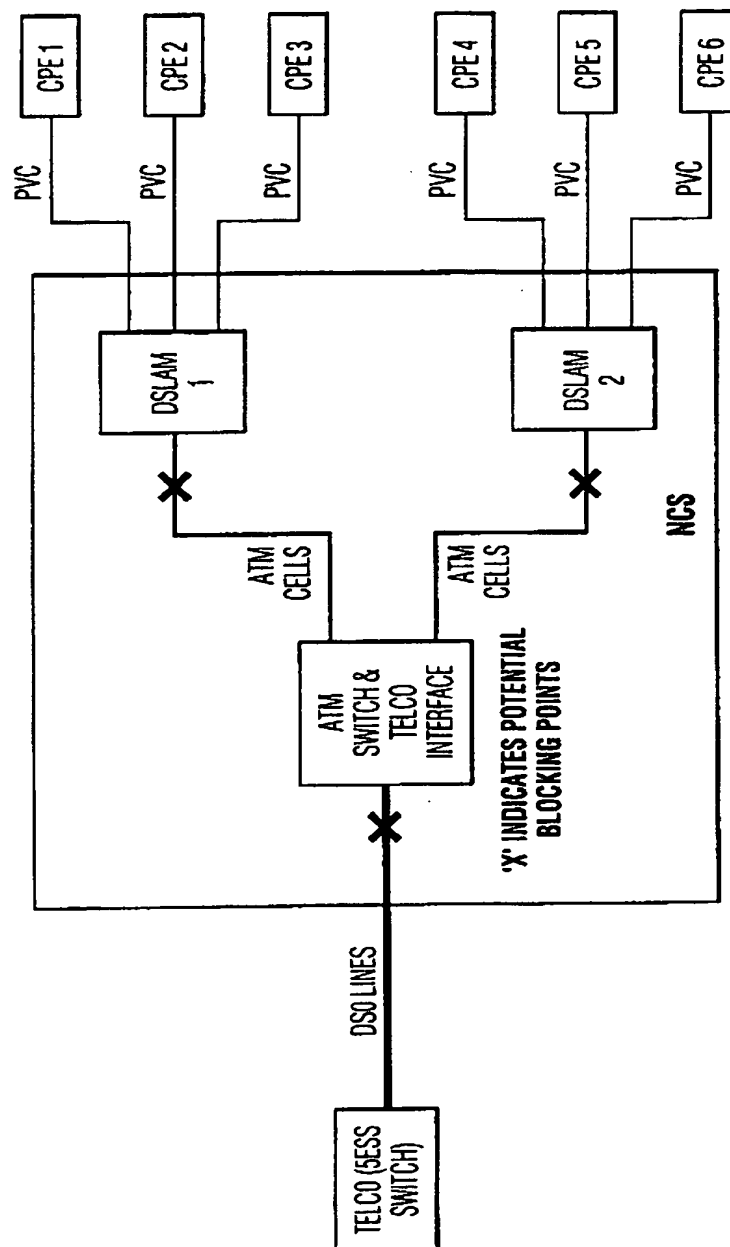


FIG. 18



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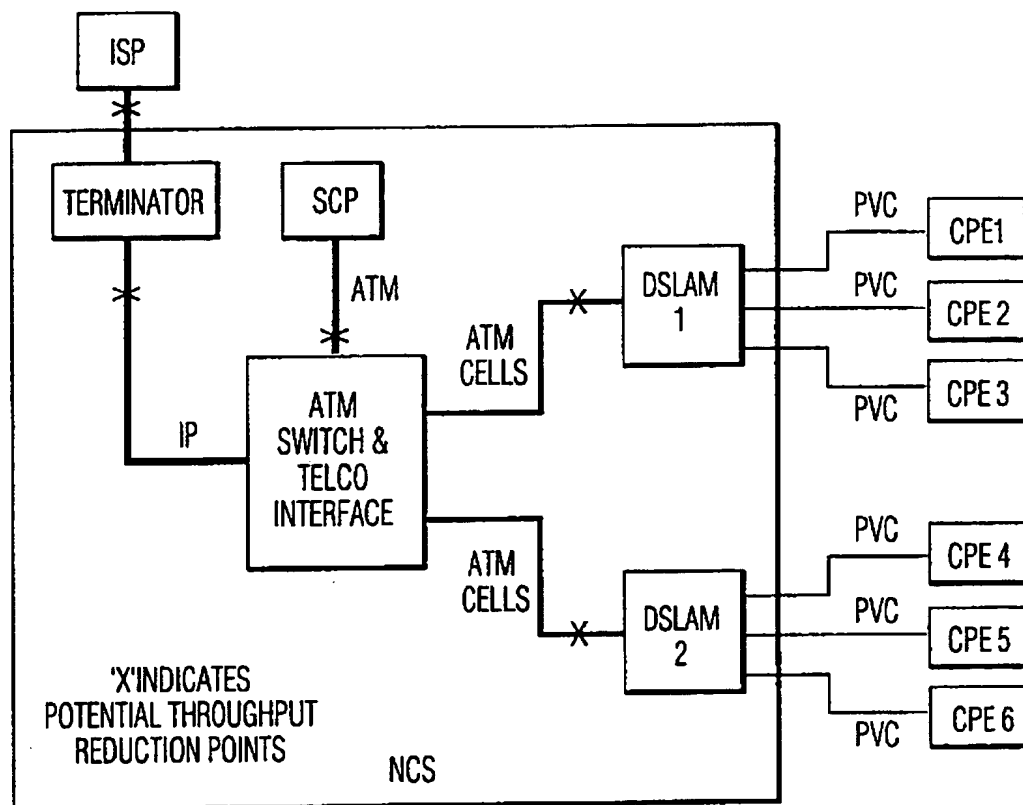


FIG. 19

# INTERNATIONAL SEARCH REPORT

Inter. Application No.

PCT/US 00/17471

**A. CLASSIFICATION OF SUBJECT MATTER**  
IPC 7 H04L12/64 H04Q11/04

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0 905 959 A (CONVERSE NETWORK SYST INC) 31 March 1999 (1999-03-31)	1-3,5,7, 9-12,14, 16,18,19 4,13,20
Y	column 12, line 7 - line 22 column 13, line 12 - line 37 claims 1-6	
Y	US 5 883 893 A (RUMER MARK ET AL) 16 March 1999 (1999-03-16) column 3, line 8 - line 15; claims 1,6,10-12 column 3, line 45 - line 51	4,13,20
A	WO 97 38511 A (AT & T CORP) 16 October 1997 (1997-10-16) page 6, line 22 -page 7, line 9 page 14, line 5 - line 30	1-20
	-/-	



Further documents are listed in the continuation of box C.



Patent family members are listed in annex.

\* Special categories of cited documents:

- \*A\* document defining the general state of the art which is not considered to be of particular relevance
- \*E\* earlier document but published on or after the international filing date
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Date of the actual completion of the international search

30 October 2000

Date of mailing of the international search report

07/11/2000

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# INTERNATIONAL SEARCH REPORT

International Application No  
PCT/US 00/17471

## C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	<p>IWATA H ET AL: "DESIGN AND IMPLEMENTATION OF AN ATM-SLIC FOR VOICE AND TELEPHONY OVER ATM(VTOA)"  IEEE VEHICULAR TECHNOLOGY CONFERENCE,US,NEW YORK, IEEE, vol. CONF. 47,  18 November 1996 (1996-11-18), pages 1375-1379, XP000741651  ISBN: 0-7803-3660-7  page 1378; table 1</p>	5,14

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Information on patent family members

International Application No

PCT/US 00/17471

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